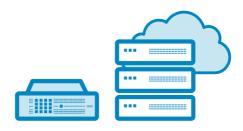
COMtrexx

Advanced Information



Information on Release 1.10	21
About these instructions	22
Gender note	23
Copyright	24
Trademarks	25
The latest information	26
Use cases / Best Practice	27
Examples from industry	28
Call centre	29
Company branches	31
Construction	33
Co-working space	34
Doctor's surgery	35
Freight forwarding	37
Law firm	39
Nursing home	41
Supermarket	44
Tradespeople	46
Automation	48
Call acceptance and announcements	49
Corporate Identity	50
Desk sharing (shared workstations)	51
Migration	52
Mobility	53
Multiple locations	55
Working together	56
Functions and applications	59
a/b-Gateway	62

Overview of Functions	63
Preparation	66
Configuration Overview	68
Configuring Providers	69
Configuring Subscribers	71
Configuring Users	73
Configuring Telephones	74
Configuring Accounts	76
Access authorisations	78
Announcement before answering	81
Automatic reception	83
Backup	87
Block- and allowlist	92
Busy-on-Busy	95
Call data	96
Call distribution	98
Call forwarding	99
Overview of Functions	100
Call Forwarding for users	102
Call Forwarding for groups	104
Call Forwarding for external numbers	106
Call on hold	108
Call Strategy	110
Overview of Functions	111
Examples	114
Call Through	121
Call waiting	123
Central caller list	124
Certificates	126
CLIP texts	129
Conference rooms	130
Configuration switchover	131
Contacts/LDAP	134

Date and Time	138
DECT system	140
Dialled calls	146
Do-not-Disturb	147
Door functions	148
E-mail	150
Emergency calls	152
Exchange line authorisation	159
Exchange line request	160
Exchange line routing	163
External access	167
Fax	170
Feature level	171
Firmware Update	173
FMC	176
Follow-me	179
Groups	181
Incoming Calls	184
InterCom	187
Internal numbers	189
LAN TAPI	192
Overview of Functions	193
Installing TAPI driver (TSP)	196
Licences	198
Locations	201
Loudspeaker Audio Out	203
Multi-path call forwarding	204
Music on Hold and Announcement	206
MWI	207
Network	208
Network interface (second)	213
Number presentation	215
Phone number display (CLIP)	217

Pick-Up and Call Pick-Up	218
Pick-up of all calls to group members	219
Powering Down and Switching Off	220
Preferred exchange line	221
Preliminary pick-up query and monitoring	222
Profiles	224
Provisioning	226
Overview of Functions	227
Provisioning files	233
Creating provisioning file	235
Query and Transfer	237
Relays (actuators)	241
Overview of Functions	242
Relay Operating Mode	244
Reserved call channels	247
Restart	248
Roaming User	250
Service data	253
Short-code number	257
Softphone	260
Switch input	263
System messages	264
Templates	266
Time control	268
Tones	269
User	272
Voice mail and fax boxes	275
Overview of Functions	276
Voice mailbox	277
Fax box	280
Fax Transfer	281
Overview of Functions	282
Installing fax printer driver on a PC	283

Configuring fax printer driver	284
Send Fax to a Recipient	286
Send Fax to several Recipients	287
VoIP	289
VoIP channels	290
External Internet Telephony	293
Internal IP telephony	295
External Private Branch Exchanges	297
VPN	298
Using the web interface	299
Basic Knowledge	300
Requirements and Notes	
Open the Web Interface	
As an administrator	303
Operating Instructions	304
Overview	305
Fields and User Elements	307
Display elements	309
Dashboard	310
Status information	311
Language	313
Feature level	314
Hardware	
Exchange lines	
Exchange line request/locations	
Exchange line request / Exchange line access number	
Miscellaneous	319
Locations	320
Emergency calls	322
Emergency call has priority	323
Emergency and service numbers	324
Legal emergency numbers	325

User-defined emergency numbers	326
Service numbers	328
Providers and Accounts	329
Overview	330
Account	333
Settings	334
Network Provider	335
Access data	336
Miscellaneous	338
Phone numbers (for PBX connection)	340
Phone numbers(for Point-to-Multipoint Connection)	341
Phone numbers (for PBX connection)	343
Expert configuration	344
VoIP account	345
Call forwarding general	346
Call forwarding unconditional (CFU)	349
Call Forwarding if Busy (CFB)	351
Call forwarding no reply (CFNR)	353
Provider	355
Provider	356
SIP	357
RTP	367
Settings	374
Number presentation (outgoing)	380
Evaluation of incoming SIP messages	385
Exchange line routing	388
Exchange line routing incoming	389
Block- and allowlist	390
Overview	391
Adding	393
Configure	395
Call distribution for	396
Overview	397

Configure	400
Exchange line routing outgoing	404
Overview	405
Configure	406
Block- and allowlist	407
Overview	408
Adding	410
Configure	412
Routing	413
User	416
Phone numbers	417
Overview	418
Configure	420
Basic Settings	421
Exchange line settings	425
Reachability	427
Authorisations	432
Voice mail and fax boxes	437
Settings	440
Profiles	443
Profile management	444
Profile Assignment	445
Configure	446
Exchange line settings	447
Reachability	449
Authorisations	453
Voice mailbox	456
Settings	458
Functional Overview	460
Global settings	461
Functional Overview	462
Basic Settings	463
Exchange line settings	465

Reachability	467
Authorisations	471
Voice mail and fax boxes	476
Settings	479
Groups	481
Phone numbers	482
Overview	483
Configure	486
Basic Settings	487
Exchange line settings	494
Reachability	496
Voice mail and fax boxes	503
Settings	506
Profiles	508
Profile management	509
Profile Assignment	510
Configure	511
Basic Settings	512
Exchange line settings	517
Reachability	518
Voice mail and fax boxes	525
Settings	527
Functional Overview	529
Basic Settings	530
Exchange line settings	535
Reachability	537
Voice mail and fax boxes	544
Settings	547
Voice mail and fax boxes	549
Phone numbers	550
Overview	551
Phone numbers	552
Configure	555

Box settings	556
Users / Authorisations	567
Announcements	569
Call acceptance / Announcement	572
Profiles	578
Profile management	579
Profile Assignment	580
Configure	581
Call acceptance / Announcement	582
Functional Overview	588
Call acceptance / Announcement	589
Basic Settings	593
Basic settings for all voice mail and fax boxes	594
Language files	597
Telephones	598
Assignment	599
Overview	600
Configure	605
Adding	615
Status	621
Provisioning	624
Internal Devices	627
Automatic receptions	628
Overview	629
Configure	631
Conference rooms	636
Call on hold	637
External Devices	638
Fax	639
Overview	640
Configure	642
Device wizard	646
Overview	647

	Configure	649
	Analogue Door Terminals	650
	VoIP Door Terminals	652
	IP switch relay	655
	Announcement outputs	657
	Door stations	659
	Overview	660
	Configuration of the door terminal	663
	Configuration of the relay	665
	Configuration of the door call	668
	Announcement outputs	670
	Overview	671
	Configuration of the announcement output	673
	Configuration of the relay	675
	Relays (actuators)	678
	Overview	679
	Configuration of the relay	682
ŧι	unctions	685
	Time control	686
	Configurations	687
	Automatic switching	688
	Configurations	689
	Adding	691
	Copy	692
	Switching times	693
	Holidays	695
	Calendar	696
	Adding	698
	Overview	699
	Call Through	700
	Exchange line settings	701
	Call Through user	703
	CLIP texts	704

Short-code number	705
Music on Hold/announcements	707
File management	708
Memory utilisation	709
Existing audio files (*.wav)	710
Own Files	711
System Files	714
File assignment	716
Music on hold	717
Music on hold with announcement	718
Music on Hold also during the ringing period	719
Music on hold	720
Announcement before answering	722
Automatic receptions	724
Conference Room	726
User data	728
Call data	729
Call data	730
Print options	736
Filter	737
Overview	738
Configure	740
Period	741
Direction	742
Type of billing	743
Call type	744
Specific filtering	745
Entry	746
Contacts and assignment	748
Contacts and assignment	749
Contacts	750
Contact groups and Assignment	754
Google Contact Groups	757

CSV import group	760
Edit	763
Contacts	764
E-mail addresses / Internet	767
Addresses	769
Contact Groups	771
Contact Groups	773
Overview	774
Configure Google contact group	778
Authenticate Google contact group	780
Authorisations	781
Settings	784
LDAP server	785
Transfer via LDAP Server	787
Conversion	788
Messages	789
Administration	795
Contact information	796
Administrator access	797
Interface	799
Operator and Location	800
Dealer	801
Licences	802
Available Licences	803
Initialising the system again	804
Activation code for extended functions	805
Date and Time	806
Date and Time	807
One-time configuration	808
Automatic synchronisation	809
Automatic distribution	810
Network	811
IP configuration	812

IP configuration	813
NET module	817
DHCP server settings for NET-module	821
DNS configuration	823
HTTP proxy configuration	825
Server service	827
Web server configuration	828
Provisioning	830
LAN TAPI	832
SFTP server	834
Gateways and routing	835
Internet Gateways	836
IP routing	837
VPN	839
IP block- and allowlist	840
IP blocklist	841
IP allowlist	842
Ports	843
VoIP	844
VoIP configuration	845
Presetting of the VoIP channels	846
Distribution of the VoIP channels	847
Proxy server for COMfortel SoftPhone	848
SIP	850
RTP	852
Quality of Service (QOS)	853
STUN server	854
SIPS / SRTP internal	855
SIPS / SRTP	856
Root certificate	857
PBX certificate	859
Certificates	861
Not trustworthy	862

Trustworthy	864
Certificate authorities	866
Web server	868
SIPS / SRTP internal	869
E-mail	870
Outgoing mail server	871
Logo management	875
Internationalisation	877
Overview	878
Configure	880
Service	882
External access	883
Remote programming	884
Access for CCC	885
Network Diagnosis	886
Trace	887
Ping	888
Logging	889
Maintenance	891
Backup	892
Configuration	893
Remote Backup	897
Restoration points	900
Service data	902
Update and Restart	903
Store firmware in the PBX	904
Firmware update via update server	905
Reset of the PBX (Restart)	907
Message Configuration	908
Transfer of system e-mail	909
Logs to be sent as e-mails	912
Events for system logs	914
Status Messages and Notes	915

System messages	916
Monitoring and Overviews	917
External call channel assignment	918
Dial plan	919
	•••
Short Reference Phone	
Information about the section	
Using Different Telephone Types	
Telephoning	
Incoming Calls	
Call-Waiting Calls	
Dialled calls	
Unsuccessful outgoing calls	
Query Calls	
Transferring calls	
Conference call	
Functions listed alphabetically	
Announcement before answering	
Automatic reception	
Blocklist (incoming)	
Call Data (Single Call Record)	
Call Forwarding for external numbers	940
Call Forwarding for groups	
Call Forwarding for users	951
Call on hold	954
Call Through	956
Call waiting	957
Conference rooms	958
Configuration switchover	960
Date and Time	961
Dialled calls	962
Do-not-Disturb	964
Door functions	965

Emergency Call Announcement	967
Firmware Update	969
Flash time	970
FMC	971
Follow-me	974
Force Public Exchange Call for Automatic Exchange Line Request	978
Groups	979
Identification (IP Address, Subnet Mask, Gateway, DHCP Client)	982
InterCom	986
LAN TAPI	987
Loudspeaker Audio Out	988
Multi-path call forwarding	989
Music on Hold and Announcement	993
Pick-up	995
PIN	996
Query and 3rd Call	997
Query and Transfer	999
Relays	1001
Roaming User	1005
Targeted VoIP access point	1007
Voice mailbox	1008
Allowlist (incoming)	1020
Hardware	1021
Information about the section	1022
Module	1023
LEDs	1024
Button	1028
Glossary	1032
Technical Terms and Functions	1033
Account	1036
APIPA	1037

Call Deflection	1038
Certificate	1039
Certificate Authority	1040
Client	1041
CLIP no screening	1042
Codec	1043
DDI	1044
DECT	1045
DHCP	1046
DiffServ	1047
DNS	1048
Domain	1049
Driver	1050
DSL	1051
DTMF	1052
DynDNS	1053
Echo Cancellation	1054
End device	1055
Ethernet	1056
Exchange line	1057
Private branch exchange	1058
Firewall	1059
Firmware	1060
Flash	1061
GAP	1062
Gateway	1063
Glossary	1064
HOST	1065
HTTP proxy	1066
HTTPS	1067
Hyperlink	1068
IAD	1069
IP address	1070

IPV4 address	1071
IPv6 address	1072
Jitter buffer	1073
LAN	1074
LDAP	1075
LED	1076
MAC address	1077
MSN	1078
NAT	1079
Network	1080
NTP	1081
Outbound proxy	1082
PD	1083
PIN	1084
PoE	1085
Port	1086
Prefix	1087
Proxy server	1088
PTMP connection	1089
PBX connection	1090
Central office	1091
QoS	1092
QR code	1093
Registrar	1094
External private branch exchange	1095
Router	1096
Routing	1097
Server	1098
SIP	1099
SIP port	1100
SIP-DDI (SIP Trunking)	
SMTP	
Softphone	1103

STUN	1104
Subnet mask	1105
Switch	1106
Syslog server	1107
System telephone	1108
TAPI	1109
TCP	1110
Template	1111
T-Net Functions	1112
UDP	1113
URL	1114
VoIP	1115
VoIP account	1116
VoIP provider	1117
Voucher	1118
VPN	1119
WAN	1120
Abbreviations	1121

Information on Release 1.10

New or revised features published with this release are listed here.

You will find further information about new features in the firmware and the associated documentation on the Internet under <u>Auerswald Support/FONtevo Support</u>.

Remote Backup

A backup can now be stored password-protected on an external server and also restored from it.

- Further help under Backup
- **Important:** The remote backup is only available with feature level 2 or higher. If you want to use this function, please contact your dealer or system partner.
 - Further help under Feature level

About these instructions

Gender note
Copyright
Trademarks
The latest information

Gender note

For the sake of better readability, the simultaneous use of the language forms masculine, female and diverse (m/f/d) is waived.

All personal designations apply equally to all genders.

Copyright

Auerswald GmbH & Co. KG, Vor den Grashöfen 1, 38162 Cremlingen, Germany, 2023

Trademarks

Disseminating and reproducing these instructions, or using and disclosing the contents, even in part, is permitted only with our express permission. Compensation will be demanded for any violations. All rights reserved.

Microsoft, Microsoft Hyper-V and Windows are trademarks of the Microsoft Corporation registered in the United States and/or other countries

Oracle® and Java are registered trademarks of Oracle Corporation and/or its affiliates.

VMware® is a registered trademark of VMware and/or its affiliates.

All other trademarks mentioned are the property of the relevant manufacturer.

The latest information

After a firmware update, you usually require a new version of the instructions.



You will find the up-to-date instructions on the Internet under Auerswald Support/FONtevo Support.

Use cases / Best Practice

Examples from industry
Automation
Call acceptance and announcements
Corporate Identity
Desk sharing (shared workstations)
Migration
Mobility
Multiple locations
Working together

Examples from industry

Call centre
Company branches
Construction
Co-working space
Doctor's surgery
Freight forwarding
Law firm
Nursing home
Supermarket
Tradespeople

Call centre

For call centres, the PBX provides a range of different functions for almost unlimited collaboration.

Personalised settings at every workplace

The Roaming User function enables users who, for example, work in a large office with workplaces that are not assigned to specific people (i.e. "hot desking") to access any telephone that is configured in the PBX. The user has the known access with his personal settings and authorisations.

If used in combination with the central caller list, the user is given access to all their calls (incoming and outgoing). This requires a COMfortel D-series telephone.

- Further help under Roaming User
- Further help under Central caller list
- Further help under <u>User</u>

Collaboration

The Groups function is used to assign individual users to different teams or departments and to control their availability, among other things, by logging them in or out.

The Rework time attribute can be used to specify that group members cannot be called after they have received an internal or external call (for example, so that additional work can be performed on the call). During this time, the Do-not-Disturb function is enabled for the user.

Different call strategies can be used to assign inbound calls to group members "properly".

Call centre

- Further help under Groups
- Further help under Call Strategy

The Announcement before answering text greets callers to a group with an announcement. If the line is busy, callers are placed directly in a waiting loop.

Further help under Announcement before answering

Use the Busy on Busy function to specify the maximum number of connections after which group members can no longer be contacted.

Further help under Busy-on-Busy

If a group of users is to be able to have an external call at any time, call channels can be reserved for this group.

Further help under Reserved call channels

Using pick-up you can take a call on your own phone that was originally sent to another internal telephone or to an own voice mailbox.

Further help under Pick-Up and Call Pick-Up

Everything in one device

COMfortel SoftPhone transforms a PC into a fully functioning office telephone.

Further help under Softphone

Company branches

The PBX offers a range of functions for company branches, including functions for mobility, call assignment and the use of a single PBX for different locations.

Providing telephony services to several company branches from one PBX

The PBX can manage telephones sited at different locations. Telephones that are not in the same location as the PBX must be configured as external extensions and assigned to a location. The correctly assigned location is necessary, among other things, so that the PBX can properly initiate an emergency call.

- Further help under Locations
- Further help under External Private Branch Exchanges
- Further help under Emergency calls

In order to be able to securely connect remote devices to the PBX, the PBX offers an integrated OpenVPN server.

Further help under VPN

If the CLIP no screening function is available for the account, and more than one location is involved, the phone number that is to be transferred can be freely assigned, independent of the connection in use.

Further help under Number presentation

Accessible at the company's premises and in every company branch

The integration of DECT base stations and telephones enables mobility anywhere in the building.

Further help under <u>DECT system</u>

The PBX has even more options for mobile telephony and mobile availability.

Further help under Mobility

Direct connection to nearby company branches

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under Announcement before answering
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

Construction

The PBX provides a range of functions for construction companies that enable almost unlimited mobility.

Accessible everywhere, always

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace and a mobile telephone at the same time.

Further help under User

Use the FMC function to configure a mobile telephone as the user's telephone, so it can be accessed on the mobile phone network at any time.

Further help under <u>FMC</u>

COMfortel SoftPhone Mobile transforms a mobile telephone into a fully functioning office telephone.

Further help under Softphone

The message forwarding function alerts the user when messages are received in the voice mailbox so they can return the call, at any time.

Further help under Voice mailbox

The PBX has even more options for mobile telephony and mobile availability.

Further help under Mobility

Co-working space

The PBX offers a range of functions for flexible workplace selection in coworking spaces.

Personalised settings at every workplace

The Roaming User function enables users who, for example, work in a large office with workplaces that are not assigned to specific people (i.e. "hot desking") to access any telephone that is configured in the PBX. The user has the known access with his personal settings and authorisations.

If used in combination with the central caller list, the user is given access to all their calls (incoming and outgoing). This requires a COMfortel D-series telephone.

- Further help under Roaming User
- Further help under <u>Central caller list</u>
- Further help under <u>User</u>

Everything in one device

COMfortel SoftPhone transforms a PC into a fully functioning office telephone.

Further help under Softphone

Doctor's surgery

For doctors' surgeries, the PBX provides a range of different functions to reduce the practise team's workload and improve patient comfort.

Arranging appointments, ordering prescriptions and patient information

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under <u>Announcement before answering</u>
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

To ensure that the correct information always comes at the right time, different announcements and availability can be set, for example, for holidays and times outside of surgery hours. These can then be enabled either manually or at specific times.

Further help under Configuration switchover

Announcements in the waiting room and consulting room

The InterCom function enables announcements from every internal telephone to be made to telephones that support this function, such as COMfortel D-series telephones. Alternatively, loudspeaker

Doctor's surgery

announcements can be made via a loudspeaker or an active loudspeaker unit.

- Further help under InterCom
- Further help under Loudspeaker Audio Out

Priority for emergency calls

The emergency priority circuit gives calls to the configured emergency numbers priority over all other calls. If required, a call can be interrupted to enable an emergency call.

Further help under Emergency calls

Patients at the door

The PBX makes it possible to operate a connected door terminal system and door opener from every internal telephone. When the door bell button is pressed, a signal can be sent not only to all the connected telephones, but also to external devices such as mobile telephones.

Further help under <u>Door functions</u>

Freight forwarding

For freight forwarding companies, the PBX provides a range of different functions to reduce the team's workload and provide almost unlimited mobility.

Accessible anywhere on the business premises and also while on the move

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace and a mobile telephone at the same time.

Further help under User

The integration of DECT base stations and telephones enables mobility anywhere on the company premises.

Further help under **DECT** system

Use the FMC function to configure a mobile telephone as the user's telephone, so it can be accessed on the mobile phone network at any time.

Further help under FMC

COMfortel SoftPhone Mobile transforms a mobile telephone into a fully functioning office telephone.

Further help under Softphone

The message forwarding function alerts the user when messages are received in the voice mailbox so they can return the call, at any time.

Further help under Voice mailbox

Freight forwarding

The PBX has even more options for mobile telephony and mobile availability.



Further help under Mobility

Law firm

For law firms, the PBX provides a range of functions to reduce the office team's workload and make billing easier.

Always reach the right contact person

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under <u>Announcement before answering</u>
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

Phone numbers and calls at a glance

Use the PBX to manage all your contacts and make the contacts you need available on your telephone.

Further help under Contacts/LDAP

Installing the PBX Call Assist CTI software on the PBX enables the PBX to also work together with DATEV.

Further help under LAN TAPI

The call data for calls with clients can also be accessed to keep an eye on costs.



Further help under Call data

Clients at the door

The PBX makes it possible to operate a connected door terminal system and door opener from every internal telephone. When the door bell button is pressed, a signal can be sent not only to all the connected telephones, but also to external devices such as mobile telephones.



Further help under **Door functions**

Nursing home

For nursing homes, the PBX provides a range of different functions to reduce the care team's workload and improve residents' safety.

Accessible anywhere in the building and also while on the move

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace and a mobile telephone at the same time.

Further help under User

The integration of DECT base stations and telephones enables mobility anywhere in the building.

Further help under DECT system

Use the FMC function to configure a mobile telephone as the user's telephone, so it can be accessed on the mobile phone network at any time.

Further help under FMC

COMfortel SoftPhone Mobile transforms a mobile telephone into a fully functioning office telephone.

Further help under <u>Softphone</u>

The message forwarding function alerts the user when messages are received in the voice mailbox so they can return the call, at any time.

Further help under Voice mailbox

Nursing home

The PBX has even more options for mobile telephony and mobile availability.

Further help under <u>Mobility</u>

Billing for residents

Further help under Call data

Announcements for residents and nursing home staff

The InterCom function enables announcements from every internal telephone to be made to telephones that support this function, such as COMfortel D-series telephones. Alternatively, loudspeaker announcements can be made via a loudspeaker or an active loudspeaker unit.

- Further help under InterCom
- Further help under Loudspeaker Audio Out

Priority for emergency calls

The emergency priority circuit gives calls to the configured emergency numbers priority over all other calls. If required, a call can be interrupted to enable an emergency call.

Further help under <u>Emergency calls</u>

Visitor at the door

The PBX makes it possible to operate a connected door terminal system and door opener from every internal telephone. When the door bell button

Nursing home

is pressed, a signal can be sent not only to all the connected telephones, but also to external devices such as mobile telephones.



Further help under **Door functions**

Supermarket

For supermarkets, the PBX provides a range of different functions that improve customer experience and provide almost unlimited mobility.

Accessible anywhere in the building

The integration of DECT base stations and telephones enables mobility anywhere in the building.

Further help under <u>DECT system</u>

The PBX has even more options for mobile telephony and mobile availability.

Further help under Mobility

Direct connection and customer information

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under <u>Announcement before answering</u>
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

Announcements in the sales area

Loudspeaker announcements can be made from any internal telephone via a loudspeaker or an active loudspeaker unit. These loudspeaker announcements can also be used to inform staff that callers are waiting in park positions.

- Further help under Loudspeaker Audio Out
- Further help under Call on hold

Tradespeople

The PBX offers a range of functions for tradespeople that provide the mobility they need.

Availability in your own workplace and also when with customers

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace and a mobile telephone at the same time.

Further help under User

The integration of DECT base stations and telephones enables mobility anywhere in the building.

Further help under DECT system

Use the FMC function to configure a mobile telephone as the user's telephone, so it can be accessed on the mobile phone network at any time.

Further help under <u>FMC</u>

COMfortel SoftPhone Mobile transforms a mobile telephone into a fully functioning office telephone.

Further help under Softphone

The message forwarding function alerts the user when messages are received in the voice mailbox so they can return the call, at any time.

Further help under Voice mailbox

Tradespeople

The PBX has even more options for mobile telephony and mobile availability.



Further help under Mobility

Automation

Some functions can be set differently e.g. for daytime, night-time, weekends, holidays and public holidays. These can then be enabled either manually or at specific times.

Further help under Configuration switchover

The PBX supports the connection and control of different types of relays, which take over a range of different functions on the PBX.

Further help under Relays (actuators)

Call acceptance and announcements

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under Announcement before answering
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

Corporate Identity

The E-mail function supports the use of bespoke logos and text elements such those used to adapt the e-mails used for sending voice mail messages to the company corporate image.

Further help under E-mail

The music on hold and announcements can be tailored to suit individual requirements.

Further help under Music on Hold and Announcement

COMfortel D-series telephones also support the use of bespoke ringtones and wallpaper.

Desk sharing (shared workstations)

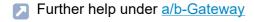
The Roaming User function enables users who, for example, work in a large office with workplaces that are not assigned to specific people (i.e. "hot desking") to access any telephone that is configured in the PBX. The user has the known access with his personal settings and authorisations.

If used in combination with the central caller list, the user is given access to all their calls (incoming and outgoing). This requires a COMfortel D-series telephone.

- Further help under Roaming User
- Further help under <u>Central caller list</u>
- Further help under <u>User</u>

Migration

Existing systems with analogue telephones can be migrated to a COMtrexx system by using the existing PBXs as a/b gateways.



Mobility

The functions described below enable the PBX to be used with mobile telephones and also with other remote telephones.

Mobile telephony

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace, a telephone that is in their home office and mobile telephone at the same time.

Further help under <u>User</u>

The integration of DECT base stations and telephones enables mobility anywhere in the building.

Further help under DECT system

Use the Call Through function to forward calls via the PBX while on the move. In this case, the person being called sees, for example, the company phone number instead of the mobile phone number.

Further help under <u>Call Through</u>

Use the FMC function to configure a mobile telephone as the user's telephone, so it can be accessed on the mobile phone network at any time. On outgoing calls, FMC acts like Call Through except that FMC is user-specific. The called person does not receive the mobile number or the central company number but the number that is also transferred from the office phone of the user.

Further help under FMC

COMfortel SoftPhone and COMfortel SoftPhone Mobile transform a PC or a mobile telephone into a fully functioning office telephone. Further help under Softphone

Staying accessible while on the move

There are a range of different forwarding functions available if staying accessible while on the move is your main concern.

Use the Call forwarding function to forward calls independently of the current situation (immediately, on busy or on no reply). Different call forwarding options are available, depending on whether calls are to be forwarded for a single user, for a group or for external numbers.

Further help under Call forwarding

Use the Follow-me function to configure forwarding from the destination telephone.

Further help under Follow-me

The Multi-Path Call Forwarding function also signals incoming calls on another telephone, even an external telephone, although it does not forward the call

Further help under Multi-path call forwarding

Multiple locations

The PBX can manage telephones sited at different locations. Telephones that are not in the same location as the PBX must be configured as external extensions and assigned to a location. The correctly assigned location is necessary, among other things, so that the PBX can properly initiate an emergency call.

- Further help under Locations
- Further help under External Private Branch Exchanges
- Further help under **Emergency calls**

In order to be able to securely connect remote devices to the PBX, the PBX offers an integrated OpenVPN server.

Further help under VPN

If the CLIP no screening function is available for the account, and more than one location is involved, the phone number that is to be transferred can be freely assigned, independent of the connection in use.

Further help under <u>Number presentation</u>

Working together

The functions described below help people work together.

Working together in teams

The Groups function is used to assign individual users to different teams or departments and to control their availability, among other things, by logging them in or out.

Different call strategies can be used to assign inbound calls to group members "properly".

- Further help under Groups
- Further help under <u>Call Strategy</u>

The Announcement before answering text greets callers to a group with an announcement. If the line is busy, callers are placed directly in a waiting loop.

Further help under Announcement before answering

Use the Busy on Busy function to specify the maximum number of connections after which group members can no longer be contacted.

Further help under Busy-on-Busy

If a group of users is to be able to have an external call at any time, call channels can be reserved for this group.

Further help under Reserved call channels

The SIP BLF function displays which team members are present.

Further help under Preliminary pick-up query and monitoring

When the PBX Call Assist CTI software application is installed on the PBX, users can see which team members are present, use video-telephony and share screens and files.

Further help under <u>LAN TAPI</u>

Picking up calls for colleagues

Using pick-up you can take a call on your own phone that was originally sent to another internal telephone or to an own voice mailbox.

Further help under Pick-Up and Call Pick-Up

If the Pick up all calls to group members function is enabled, one member of the group can not only pick up calls to the group's phone number but also direct calls to individual group members.

Further help under Pick-up of all calls to group members

The Enquiry prior to pick-up function enables users to see which call they want to receive before they pick it up.

Further help under Preliminary pick-up query and monitoring

Transferring calls to colleagues

If a caller wants to speak to a specific colleague, their call can be transferred directly to their requested call partner. If necessary, the caller can also be announced by a query call.

Further help under Query and Transfer

If monitoring and BLF are enabled, you can look at the telephone display, before you transfer the call, to find out which colleague is free to take the call.

Working together

Further help under Preliminary pick-up query and monitoring

The open query call can be used to place callers who are to be transferred in a park position so that the requested call partner can take the call on a different internal telephone.

Further help under Call on hold

Functions and applications

a/b-Gateway

Access authorisations

Announcement before answering

Automatic reception

Backup

Block- and allowlist

Busy-on-Busy

Call data

Call distribution

Call forwarding

Call on hold

Call Strategy

Call Through

Call waiting

Central caller list

Certificates

CLIP texts

Conference rooms

Configuration switchover

Contacts/LDAP

Date and Time

DECT system

Dialled calls

Do-not-Disturb

Door functions

E-mail

Emergency calls

Exchange line authorisation

Functions and applications

Exchange line request

Exchange line routing

External access

Fax

Feature level

Firmware Update

FMC

Follow-me

Groups

Incoming Calls

InterCom

Internal numbers

LAN TAPI

Licences

Locations

Loudspeaker Audio Out

Multi-path call forwarding

Music on Hold and Announcement

MWI

Network

Network interface (second)

Number presentation

Phone number display (CLIP)

Pick-Up and Call Pick-Up

Pick-up of all calls to group members

Powering Down and Switching Off

Preferred exchange line

Preliminary pick-up query and monitoring

Profiles

Provisioning

Functions and applications

Query and Transfer

Relays (actuators)

Reserved call channels

Restart

Roaming User

Service data

Short-code number

Softphone

Switch input

System messages

Templates

Time control

Tones

User

Voice mail and fax boxes

VoIP

VPN

a/b-Gateway

Overview of Functions
Preparation
Configuration Overview
Configuring Providers
Configuring Subscribers
Configuring Users
Configuring Telephones
Configuring Accounts

Overview of Functions

Existing systems with analogue telephones can be migrated to a COMtrexx system by using the existing PBXs as a/b gateways.

- Note: The following description applies to COMtrexx from firmware version 1.6 and COMmander/COMpact PBXs from firmware version 8.4.
- Note: Alternative for COMtrexx Business: By upgrading with a 4FXS module, there are now four internal ports for analogue devices (telephones, fax machines, door stations, announcement outputs) available.

PBXs supported as an a/b gateway

The following Auerswald/FONtevo PBXs can be used as an a/b gateway in a COMtrexx. The value shown in brackets is the maximum number of VoIP channels that can be used in the direction of the COMtrexx. This value depends on the extension level (modules or releases required).

- COMpact 4000 (10)
- COMpact 5000/R (14)
- COMpact 5200/R (20)
- COMpact 5500 (32)
- COMmander 6000/R/RX (38)

Supported Functions

The following functions can be used in the normal way on the analogue telephones connected via the a/b gateway.

Incoming call with phone number and name display

In an external call, the name is only displayed if the phone number has been entered as a contact in the COMtrexx.

· Outgoing call

Every call (including those to analogue devices) is sent via the COMtrexx.

Transfer with an announcement

The second VoIP channel on the a/b gateway is always used to transfer the call.

· Splitting and targeted disconnection

Analogue telephones connected to the a/b gateway must be used in a different way to perform the following functions.

 Use of function codes (e.g. for pick-up, logging into/out of groups, do not disturb)

This requires an exchange line access with ** account number

Example of Do not disturb on, with account number 9201: ** 9201 ##

8* 21 1

Telephones connected to the COMtrexx can be used in the normal way to perform the following functions, even when used together with the analogue telephones.

- Incoming calls from analogue telephones with phone numbers and name display
- Outgoing calls to analogue telephones
- BLF status of the analogue telephones (free, call, busy)

Unsupported Functions

The following functions cannot be used:

- · Cross-system busy statuses
- Transfer without announcement (blind transfer)
- · Three-way call
- Address books of the PBX being used as an a/b gateway
- Provisioning by the COMtrexx of the PBX being used as the a/b gateway (or vice versa)

Preparation

The following preparations must be made in advance to ensure configuration runs smoothly.

Delete settings that are no longer required

Delete the following settings in the PBX that is to be used as the a/b gateway because they are no longer required.

- Provider
- Accounts
- Subscriber
- · Address books

Creating a telephone numbering plan

The same internal phone numbers for the analogue telephones must be used in both PBXs. To achieve this, first create a valid telephone numbering plan for both PBXs.

Note the following points for the internal phone numbers for analogue telephones:

- The phone numbers in the following components must be identical
 - in the COMtrexx: User
 - in the PBX used as the a/b gateway: Subscriber, account MSN
- · Phone numbers must have at least 3 digits
 - Note: An additional telephone number range for account numbers must be added to the telephone numbering plan of the PBX to

Preparation

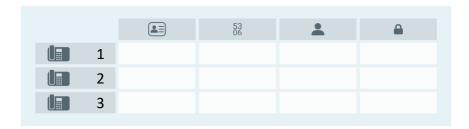
be used as an a/b gateway. Ideally this should be composed as follows: digit + internal number of each telephone.

- Example:

- Unused telephone number range: 9...
- Internal numbers of the analogue telephones: 201 211
- Account numbers: 9201 9211

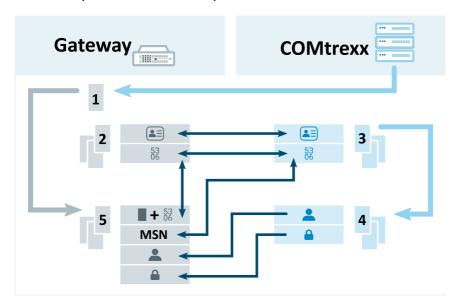
A table in which the following values can be stored temporarily may be useful during configuration (especially for larger installations):

- Name
- Internal number
- Username
- Password



Configuration Overview

This overview shows the components required in both PBXs and the value relationships between the components.



- 1. **Provider** for the connection to the COMtrexx
- 2. Subscriber for every analogue telephone
- 3. **User** for every analogue telephone
- 4. Phone for every analogue telephone
- 5. Account for every analogue telephone
 - Details about settings under <u>Configuring Providers</u>, <u>Configuring Subscribers</u>, <u>Configuring Users</u>, <u>Configuring Telephones</u>, <u>Configuring Accounts</u>

Configuring Providers

Requirements:

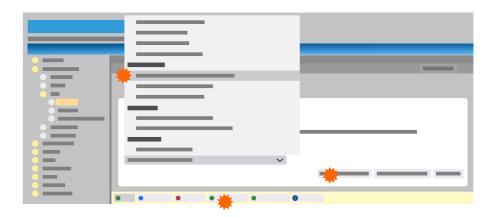
You need to know the IP address and port of the COMtrexx

Configuration location:

- The PBX system used as the a/b gateway
 - Public switched tel. networks > VoIP > Provider

Components to be configured:

- A provider based on the following template: International > int Auerswald PBX sub system IPv4 V201
 - Import > Selection of the template > Import provider



Make the following settings for the provider:

- Configure
 - SIP > Domain: COMtrexx IP address

- SIP > Registrar: IP address and port of the COMtrexx
- Settings > Sub-system operation: on
- Settings > The provider has emergency call ability: on
- Number presentation (outgoing) > Format of called phone number: send as dialled
- Number presentation (outgoing) > Format of own phone number: Without country code (e. g. 05306...)
- Number presentation (outgoing) > Number presentation type:
 In the display text
- Number presentation (outgoing) > Method of number presentation suppression: Anonymous

Configuring Subscribers

Requirements:

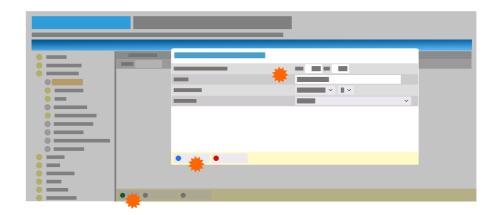
- Telephone numbering plan name and internal number for every analogue telephone
 - Further help under Preparation

Configuration location:

- The PBX system used as the a/b gateway
 - Subscriber (scr.) > Phone numbers

Components to be configured:

- One subscriber for every analogue telephone, with name and internal number from the telephone numbering plan:
 - Further help under Preparation
 - New > Settings under Add subscriber > Save



Make the following settings for the subscriber:

Configure

- Module/Port: Module/port to which the telephone is connected
- Exchange line settings > Exchange line authorisation:
 International
- Exchange line settings > Preferred exchange line: Automatic occupation
- Authorisations > Transfer of external calls to external; on
- Settings > Exchange line request: Direct exchange line phone
- Settings > Transmission of the display name: on
- Note: Properties such as the exchange line authorisation for the analogue telephone must be set via the user of the COMtrexx, not via the subscriber.

Configuring Users

Requirements:

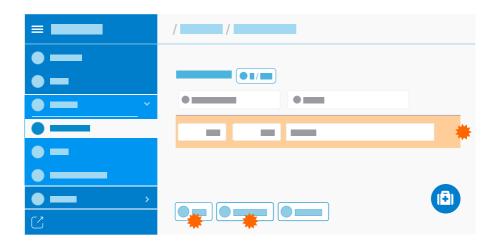
- Telephone numbering plan name and internal number for every analogue telephone
 - Further help under Preparation

Configuration location:

- COMtrexx
 - User > Phone numbers

Components to be configured:

- One user for every analogue telephone, with name and internal number from the telephone numbering plan:
 - Further help under Preparation
 - New > Settings > Save



Configuring Telephones

Requirements:

- · Configured users
 - Details about settings under Configuring Users

Configuration location:

- COMtrexx
 - Telephones > Assignment

Components to be configured:

- One telephone for every analogue telephone:
 - New > settings under Adding a device > Save
- Make the following settings for the telephone:
 - Name: Name
 - Type: SIP soft client
 - Manufacturer: Another manufacturer (not necessary for firmware versions prior to 1.6.0)
 - Important: Do not use the name Auerswald.
 - **User**: The user configured for the telephone

Configuring Telephones



Telephone data required for further use:

- /
 - SIP user name



Configuring Accounts

Requirements:

- A telephone numbering plan containing an internal number and name for every analogue telephone
 - Further help under Preparation
- · A configured provider
 - Details about settings under Configuring Providers
- You need to know the SIP user name and Password of the telephones configured in the COMtrexx
 - Details about settings under <u>Configuring Telephones</u>

Configuration location:

- The PBX system used as the a/b gateway
 - Public switched tel. networks > VoIP > Accounts

Components to be configured:

- · One account for every analogue telephone:
 - New > settings under Adding a device > Save
- · Make the following settings for the account:
 - Provider: int Auerswald PBX sub system IPv4 V201
 - Account name: Name
 - Connection type: PTMP connection
 - Use: on



The following additional settings must be made for the account:

Configure

- Network Provider > Account has emergency call ability: off
- Access data > Exchange line access number (account number): Digit in the free telephone numbering range + internal number from the telephone numbering plan
 - Further help under Preparation
- Access data > Username: SIP user name of the telephone configured in the COMtrexx
- Access data > Password: Password of the telephone configured in the COMtrexx
- Phone Numbers for PTMP connection > Multiple subscriber number (MSN): internal number from the telephone numbering plan
 - Further help under Preparation

Access authorisations

In order to prevent important settings from being changed accidentally or by unauthorised people, the PBX can be protected from unauthorised access.

Administrator

The administrator has unrestricted access to the web interface (access with the admin user name and admin password). The PBX can completely be configured and the PINS and passwords of all internal users can be changed with this access.

Users (internal users)

The user is any internal user of the PBX. For each user, a user PIN is generated. It is used, among other things, to access the voice mail and fax box and for the functions FMC and Roaming User.

Additional app passwords for users

PBX access data must be saved frequently in external systems. This will lead to security risks, if these systems are insufficiently secured.

Therefore, additional user passwords for accessing the web interface or API can be generated (currently not used).

If an app password has been generated for the user and the corresponding access authorisation has been enabled, the previous user password loses the access authorisation for the corresponding application.

Configuration

- · Configuring administrator access.
 - Administration > Contact information > Administrator access
- Change the automatically generated user PINs.
 - User > Phone numbers > / > Basic Settings
- Generating additional app passwords (web interface/API) for a user.
 - User > Phone numbers > / > Basic Settings
- Configuring authorisation for controlling of the PBX via phone.
 - User > Phone numbers > / > Authorisations

Password

Password that consists of at least 8 characters containing digits, letters (upper and lower case, but no German umlauts and β) and special characters – β . ! α * ' β () δ = + \$, .

Specific characteristic of the Admin password: All characters can be used without restrictions.

Passwords with a low security level cannot be saved. A high level does not necessarily indicate a secure password.

Note: General rule: The more complex the password, the shorter it can be. The longer the password, the simpler the character strings used can be. Coloured bars provide direct visual feedback about the password strength when a password is entered.

▲ Caution: Visible PINs and passwords are a security risk.

PIN

The PIN consists of 6 digits.

All the PINs in the PBX are unique. For this reason, you cannot assign the same PIN twice.

Do not use dates of birth, dates or easy-to-guess PINs like 1111111 or 123456.

Setting a user PIN is not absolutely necessary.



▲ Caution: Visible PINs and passwords are a security risk.

Announcement before answering

This function announcement before answering allows a greeting message to be played to the caller before accepting the call and/or if all group members are busy. If busy, callers are then put on hold in the waiting loop.

The Busy-on-Busy function can be used to limit the number of calls put on hold. This prevents more callers from being connected to the waiting loop than can be answered in the available time. Waiting callers are held for a maximum of 8 minutes.



Further help under Busy-on-Busy

Process

Calls are accepted with announcement before answering in the following way:

- 1. The call is accepted by the PBX (also if busy).
- 2. Callers hear an announcement while the free group members are called

The following rules/restrictions apply:

- The external caller is charged.
- Optionally, free group members can be called after the announcement.
- If all the group members are busy, callers are put on hold after the announcement
- 3. As soon as one of the called group members lifts the handset or ends the call beforehand in the event of a busy call, they are connected to the person on hold.

The following rules/restrictions apply:

- Waiting calls are put on hold by the PBX for a maximum of 8
 minutes after the announcement before answering. If the call is not
 taken within these 8 minutes, it is disconnected.
- The on hold time is reduced if internal phones are ringing but not taking the call. Then waiting callers are turned away after less than 8 minutes.

Configuration

- Managing and assigning audio files for announcements. If required, switching on music on hold also during the ringing period.
 - Functions > Music on Hold/announcements
- Configuring announcement before answering separately for groups. If required, additionnaly configuring Busy-on-Busy.
 - Groups > Phone number > / > Reachability
- · Configuring announcement before answering configuration-dependent.
 - Groups > Profiles > / > Reachability

Alternative function on an internal phone:

- · Recording, listening to and deleting announcements.
 - Description of operation under <u>Announcement before answering</u>

Automatic reception

This function automatically connects an external caller with the person they want to contact. The caller hears an announcement (e.g. "... If you would like to talk to the Sales team, press 1 ...") and if they then redial, they can call a specific internal user, a group, a voice mailbox, a different automatic reception, or a user whose number has been diverted to an external number. Depending on the configuration, the caller either redials using a DTMF suffix dialling digit of between 0 and 9 or by dialling the entire internal number.

A total of ten receptions can be set up. These can either be switched in parallel or in series. As a result, several automatic receptions can be operated in parallel, for example in multi-company operation or, if required, one main reception with a number of sub-receptions can be set up.

Process

Calls are accepted in the automatic reception in the following way:

- 1. A caller reaches the automatic reception (e.g. via the external call distribution function).
- 2. The caller hears a greeting, which might also be repeated and they then have the option of selecting a call destination by using a single number or a phone number for DTMF suffix dialling. They can also redial during the announcement.

The following rules/restrictions apply:

 The announcement is always played to the caller when their call is received. For this reason, the caller might hear the call tone for a few seconds before their call is accepted, for example, if several callers are listening to announcements at the same time.

- The user is given a maximum of 10 seconds in which to enter each digit. If this time period is exceeded, or an invalid phone number is dialled, the announcement is played when the destination is busy/ does not reply and the caller will be automatically connected with the automatic reception they called previously.
- External calls can only be transferred over the automatic reception if the provider does not block DTMF code presentation.
- If an external caller is transferred to the automatic reception by a query call from an internal telephone, the person transferring them hears the announcement and can then select the destination themselves. After that, the caller is connected with the selected destination. If the person transferring the calls hangs up before selecting a destination, the caller hears the announcement and can select a destination themselves.
- 3. The call is transferred to the destination after a valid number is entered.

The following rules/restrictions apply:

- When a call is transmitted via the automatic reception, the caller's phone number is transferred (not the reception's number).
- If the call is not answered, the announcement if the destination is busy/does not reply is played. In this case, the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").
- If the destination number is busy, the call waiting function is enabled if this function is enabled for the destination. If not so or the destination rejects the call waiting call, the announcement if the destination is busy/does not reply is played. In this case,

the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").

- If the caller dials an invalid number, the announcement if the destination is busy/does not reply is played. In this case, the call is then either ended, the default destination is called, or the caller is reconnected to the automatic reception previously called (depending on the setting for the system "Reaction if the destination is busy or does not reply").
- If the caller does not dial a number, they are transferred to the announcements set for the default destination number.
- The block- and allowlists and Do-not-Disturb settings that apply to in-house call destinations are taken into account when the call is transferred.
- Further help under <u>Block- and allowlist</u>
- Further help under <u>Do-not-Disturb</u>

- Managing and assigning audio files for announcements.
 - Functions > Music on Hold/announcements
 - Note: Additionally, announcements for announcements before answering can be used for automatic receptions.
- · Creating and configuring automatic receptions.
 - Internal Devices > Automatic receptions
- · Setting up call distribution for automatic receptions.

Exchange lines > Exchange line routing > Exchange line routing incoming

Alternative function on an internal phone:

- Recording, listening to and deleting announcements.
- Description of operation under <u>Automatic reception</u>
- Description of operation under <u>Announcement before answering</u>

Backup

The following options are available for backing up the configuration data of the PBX:

- · Configuration file *.cfg on a data carrier
- Restoration point in the PBX
- Backup on an external server (remote backup)

In all three cases, there is the option of restoring the configuration data without any network settings.

When creating the configuration file *.cfg, the data packages PBX data, contact images as well as music-on-hold and announcements are also optional.

Remote Backup

A backup can be stored password-protected on an external server and also restored from it.

- Important: The remote backup is only available with feature level 2 or higher. If you want to use this function, please contact your dealer or system partner.
 - Further help under <u>Feature level</u>
- Important: If you want to use the remote backup server offered by Auerswald, please contact your dealer or system partner.
- Information on using your own remote backup server can be found in the Auerswald DokuWiki.

Legal notes

Important: Note the following legal notes before saving the backup.

The backup includes, amongst others, the following legally especially protected components:

- · Configuration of the PBX
- · Names of internal users
- · Passwords of PBX, users, VoIP accounts and VoIP users
- Address books of the PBX
- Connection data sets with destination, source, and time information of a telecommunication connection

Before the backup is being saved, all users to the PBX must be informed that these contents will be forwarded and about who will use the data received. It is essential to store the data securely and in accordance with the legal requirements. You must ensure that all legal requirements are met if you want to supply the data backup to your specialised dealer or the manufacturer for error analysis.

Accept the legal notes by entering "ok" in the entry field.

- Saving and reading out the configuration data on and from a data carrier
 - Administration > Maintenance > Backup > Configuration
- Delete current configuration and start with a new configuration
 - Administration > Maintenance > Backup > Configuration
- Regenerate configuration

- Administration > Maintenance > Backup > Configuration
- Saving and reading out the configuration data on and from an external server
 - Administration > Maintenance > Backup > Remote Backup
- Set up and assign own remote backup server.
 - Administration > Maintenance > Backup > Remote Backup
 - Information on using your own remote backup server can be found in the Auerswald DokuWiki.
- · Inserting and restoring from restoration points
 - Administration > Maintenance > Backup > Restoration points
- · Automatic upload
 - Using the PBX (only COMtrexx Business)

Alternative function by pressing a key on the PBX (only COMtrexx Business)

- Restoring factory settings
- Description of operation under Button

Storing the Configuration for Automatic Upload on a USB Memory Stick

Requirements:

- PBX to be configured: COMtrexx Business
- Configuration file saved on the PC
 - Note: Network settings in the configuration file are adopted during automatic loading. If these settings were not saved in the

configuration file, the network settings of the factory settings are kept during automatic loading.

- 1. Name the configuration file saved on the PC auconfig.cfg.
- 2. Save the configuration file in the root directory of a USB memory stick.

Configuration of Automatic upload

Requirements:

- · COMtrexx Business in the delivery state
- On a USB memory stick saved configuration file with the name auconfig.cfg
- Warning: Touching voltage-carrying conductors or telephone connections can cause life-threatening electric shocks.
 - For some installation and maintenance work, it is necessary to open the PBX while it is in operation (this should only be done by a qualified electrician). Make sure that the PBX is never left unattended while it is running with an open casing. Close the casing when installation has been completed.
 - Only qualified electricians should perform installation work within an open housing or perform service tasks using the buttons inside the casing. If necessary, commission an authorised dealer to perform this work.
- Warning: Power surges, which may occur during electrical storms, can cause life-threatening electric shocks, or damage or destroy the PBX.
 - Do not open the PBX during an electrical storm.
- 1. Switch off the PBX.

- Insert the USB memory stick into the USB socket (USB host) on the CPU module.
- 3. Switch on the PBX.

The configuration data saved on the USB memory stick are automatically loaded into the PBX. The PBX afterwards.

If the **Power** LED lights up in green again, the restart is complete and the PBX is ready for operation.

- Note: If the Power LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.
- Note: Network settings in the configuration file are adopted during automatic loading. If these network settings were not saved in the configuration file, the network settings of the factory settings are kept.

Block- and allowlist

Block- and allowlists are created in the exchange line routing for systemwide usage and are assigned to users, groups or special functions. The PBX distinguishes between two types of lists:

Blocklist incoming

If you do not want to be reached by certain numbers or if you want to prevent callers from reaching a remote extension, their external numbers and/or anonymous callers can be assigned to a blocklist for incoming calls.

Allowlist incoming

If you still want to allow specific people to call you despite enabled Do-not-Disturb function, their external numbers can be assigned to an allowlist for incoming calls in the PBX.

Further help under <u>Do-not-Disturb</u>

Blocklist outgoing

To prevent that users or groups call certain phone numbers, these numbers can be entered in the blocklist for outgoing calls.

Further help under <u>Exchange line routing</u>

Allowlist outgoing

Phone numbers that are on the allowlist for outgoing calls unblocks individual phone numbers/number ranges blocked by the exchange line routing.

Further help under Exchange line routing

- Creating the block- and allowlist (incoming).
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Block- and allowlist
- Creating, enabling and disabling the block- and allowlist for exchange lines.
 - Exchange lines > Exchange line routing > Exchange line routing outgoing > / > Block- and allowlist
- Enabling the block- and allowlist (incoming) separately for users/ groups.
 - User > Phone numbers > / > Reachability
 - Groups > Phone numbers > / > Reachability
- · Assigning exchange line routing with configured block- and/or allowlist (outgoing) separately for users/groups.
 - User > Phone numbers > / > Exchange line settings > **Exchange line routing**
 - Groups > Phone numbers > / > Exchange line settings > **Exchange line routing**
- Configuring the block- and allowlist (incoming) for users/groups configuration-dependent.
 - User > Profiles > / > Reachability
 - Groups > Profiles > / > Reachability

- Assigning exchange line routing with configured block- and/or allowlist (outgoing) for users/groups configuration-dependent.
 - User > Profiles > / > Exchange line settings > Exchange line routing
 - Groups > Profiles > / > Exchange line settings > Exchange line routing

Alternative functions on an internal phone:

- Enabling and disabling the blocklist (incoming) for users.
 - Description of operation under <u>Blocklist (incoming)</u>
- Enabling and disabling the allowlist (incoming) for users.
 - Description of operation under Allowlist (incoming)

Busy-on-Busy

With the Busy-on-Busy function, calls to a group can be limited in such a way that they can be meaningfully handled by the logged-in group members. If the Busy-on-Busy limitation takes effect, the group is no longer accessible. Other callers will then hear the busy tone. Call forwardings and multi-path call forwardings are still possible.

If the announcement before answering is enabled at the same time, the Busy-on-Busy function can be used to limit the number of calls put on hold in the waiting loop. This prevents more callers from being connected to the waiting loop than can be answered in the available time. Waiting callers are held for a maximum of 8 minutes

If the announcement before answering is disabled, Busy-on-Busy should be switched on dynamically depending on the available group members.



Further help under <u>Announcement before answering</u>

- Configuring Busy-on-Busy separately for groups.
 - Groups > Phone number > / > Reachability
- Configuring Busy-on-Busy configuration-dependent/time-controlled.
 - Groups > Profiles > / > Reachability

Call data

The call data memory of the PBX, can store up to 18,000 call data records and it is secured against power failures.

Contents of the Call Data Record

- · Date at the beginning of the call
- Time at the beginning of the call
- Duration of the call
- Telephone number of the external communication partner if known
- Name of the external communication partner if known
- Phone number of the internal phone (No. invoice+No. real)
- Name of the internal phone (Name invoice+Name real)
- Own external phone number (**Connection no.**)
- Call direction (outgoing or incoming)
- · Type of billing
- · Call/type of call

- · Configuring call data management.
 - User data > Call data > Entry
- · Configure the columns of the call data list.
 - User data > Call data > Print options
- · Configuring filter.

Call data

- User data > Call data > Filter
- · Managing call data.
 - User data > Call data > Call data

Alternative function on an internal phone:

- · Deleting call data records.
 - Description of operation under <u>Call Data (Single Call Record)</u>

Call distribution

The call distribution defines the internal numbers to which incoming external calls are signalled, e.g. which internal phones ring when you receive a call. This is determined in the exchange line routing.



Further help under **Exchange line routing**

Call forwarding

Overview of Functions
Call Forwarding for users
Call Forwarding for groups
Call Forwarding for external numbers

Overview of Functions

Call forwarding allows calls to be forwarded, e.g. to a smartphone.

The PBX supports three different call forwarding types:

- Call Forwarding for users
- · Call Forwarding for groups
- · Call Forwarding for external numbers

As there can be many different reasons for not accepting a call, for example, the user in question is not present or is on another line, three call forwarding types are provided:

immediately: The phone being called does not ring. The call is forwarded immediately.

on busy: The call is forwarded immediately but only if the phone being called is busy.

on no reply: The phone being called rings for a certain time. If the call is not accepted, it is forwarded.

- Note: A different destination can be configured for each call forwarding type.
- Note: If Enable Call Forward Busy and Call forwarding on no reply are enabled at the same time, both types apply. Depending on which case occurs the telephone is busy or nobody answers it the call is forwarded to their respective telephone numbers.
- Note: If Call forwarding immediately is enabled in addition to Enable Call Forward Busy and/or Call forwarding on no reply, all of the calls are forwarded immediately. In this case, the other call forwarding types are overridden, but remain enabled. As soon as Call forwarding

Overview of Functions

immediately is disabled, the other types that are still enabled are again applicable.

Call Forwarding for users

When call forwarding for users is used, internal and external calls to single users can be rerouted to other internal phone numbers or external connections. In this way, the user in question or their substitute can take these calls on another phone.

Call forwarding for users makes sure that the user in question or their substitute is always available at his own phone number, even if the user cannot take a call on his own phone.

For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply to the user are checked before a call is forwarded.

- Further help under **Exchange line routing**
- Note: 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 a VoIP network provider supporting call deflection. 2. The rerouted phone number is neither called as a group member, nor is it required to consider parallel calling.

- Enabling the following authorisations separately for the involved internal users: Controlling of the PBX via phone, Configuring CF (sub) to ext. number
 - User > Phone numbers > / > Authorisations

- Enabling the authorisations for the involved internal users configuration-dependent/time-controlled:
 - User > Profiles > / > Authorisations
- Configuring, enabling call forwarding separately for users and making the following additional settings: only external calls, CF on group call
 - User > Phone numbers > / > Reachability
- Configuring, enabling call forwarding configuration-dependent/timecontrolled for users and making the additional settings.
 - User > Profiles > / > Reachability

Alternative functions on an internal phone

- Note: For configuring using a character string on an internal phone, the user needs an authorisation for controlling of the PBX.
 - Configuring, enabling and disabling unconditional call forwardig
 - Configuring, enabling and disabling call forwardig if busy
 - Configuring, enabling and disabling call forwardig on no reply
 - Configuring and enabling call forwarding on an analogue T-Net phone via menu/T-Net function key (see phone manual)
 - Description of operation under <u>Call Forwarding for users</u>

Call Forwarding for groups

When call forwarding for groups is used, internal and external calls to a group can be rerouted to other internal telephones or external connections. In this way, a person that is not a member of this group and therefore cannot log in, can receive these calls.

Call forwarding for groups makes sure that someone is always available on the group number, even if the calls cannot be taken by a member of the group.

For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply for the group are proved before a call forwarding is executed.

- Further help under **Exchange line routing**
- Note: 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 a VoIP network provider supporting call deflection. 2. The rerouted phone number is neither called as a group member, nor is it required to consider parallel calling or fallback on busy.

- Enabling the following authorisations separately for the involved internal users: CF (groups) / Follow-me, Controlling of the PBX via phone, Configuring CF (sub) to ext. number
 - User > Phone numbers > / > Authorisations

- Enabling the authorisations for the involved internal users configuration-dependent/time-controlled:
 - User > Profiles > / > Authorisations
- Configuring, enabling call forwarding separately for groups and making the following additional settings: only external calls, CF on group call
 - Groups > Phone numbers > / > Reachability
- Configuring, enabling call forwarding configuration-dependent/timecontrolled for groups and making the additional settings.
 - Groups > Profiles > / > Reachability

Alternative functions on an internal phone

- Note: For configuring using a character string on an internal phone, the user needs an authorisation for controlling of the PBX.
 - Configuring, enabling and disabling unconditional call forwardig
 - Configuring, enabling and disabling call forwardig if busy
 - Configuring, enabling and disabling call forwardig on no reply
 - Description of operation under <u>Call Forwarding for groups</u>

Call Forwarding for external numbers

When call forwarding for external numbers is used, calls directed to an external number (dialled by an external caller) are rerouted to other external connections. In this way, these calls can be taken on another telephone, for example, on a smartphone if no one can take the call internally.

- Note: If you want to also forward internal calls, use call forwarding for users or for groups instead of call forwarding for external numbers.

 This also allows calls to be forwarded to internal telephones.
- Note: The function can be configured with a VoIP exchange line via keypad codes (depending on the provider) via the keypad of the phone.

- Enabling the following authorisation separately for the involved internal phones: Features to be configured by the network provider (CF, key pad)
 - User > Phone numbers > / > Authorisations
- Enabling the authorisation for the involved internal phones configuration-dependent/time-controlled:
 - User > Profiles > / > Authorisations
- Configuring and enabling call forwarding (if necessary, configurationdependent/time-controlled)
 - Exchange lines > Exchange line routing > Call distribution for
- Making additional settings for call forwarding separately for providers.
 - Exchange lines > Providers and Accounts > / > Settings

- · Making additional settings for call forwarding separately for accounts.
 - Exchange lines > Providers and Accounts > / > Settings >Miscellaneous > Expert configuration

Alternative functions on an internal phone

- · Configuring, enabling and disabling unconditional call forwardig
- Configuring, enabling and disabling call forwardig if busy
- · Configuring, enabling and disabling call forwardig on no reply
 - Description of operation under <u>Call Forwarding for external</u> numbers

Call on hold

Call parking lets a caller in a park position be transferred so that the desired communication partner can accept the call on another internal phone.

Process

1. The call is first transferred to a phone number that has been previously specified for this function and then parked.

The following rules/restrictions apply:

- Having dialled the park position, you hear the call tone and you
 can hang up. If there is a busy tone after having dialled the park
 position, this position is already in use. Press the Flash key or the
 Query key to get back and dial again the base phone number with
 a different park position.
- The user assigned to the phone needs an authorisation for call parking.
- 2. The caller is kept on hold in the PBX and listens to music on hold.
- 3. The desired call partner can accept the call from a different phone by dialling the phone number to which the call has been transferred before.

The following rules/restrictions apply:

- The user assigned to the phone needs an authorisation for call parking.
- Description of operation under <u>Call on hold</u>

- · Making settings for call parking.
 - Internal Devices > Call on hold
- Switching on the authorisation for call parking for the involved internal phones.
 - User > Phone numbers > / > Authorisations

Call Strategy

Overview of Functions Examples

Overview of Functions

Different call strategies can be set, as it does not always make sense that all phones logged in in a group (e.g. in hotline mode) ring at the same time.

The following call strategies are possible:

All

- All users logged into the group ring simultaneously (except fixed phones)
- Call waiting is applied to all logged-in phones when they are busy
- The call is ended by the network provider (after approx. 2 minutes)
- In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time

Linear

- All phones logged into a group ring directly one after the other
- · Only one phone rings at a time
- · Busy users are skipped
- In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time

Adding

- All users logged into a group ring one after the other delayed by the specified time
- · Busy users are skipped

 In the case of call forwarding on no reply, the group call is forwarded to the destination already after the configured delay time

The following properties can additionally be selected for the call strategy, whereby the call strategy **All** restricts the selection for further properties.

fixed	The order of the called phones depends on the assigned priority.
balancing	The order of the called phones depends on a dynamic priority list. After each call, the phone involved is placed last.
rotating	After the first loop, another one starts. In the second loop, call waiting is applied to the busy users. The call is ended by the network provider (after approx. 2 minutes).
without rotating	After the first loop, the call is disconnected.
Split Group	During the call phase, the group needs to be reachable for further callers. The group is split up so that as many users as possible are available for each call.
without Spli Group	A second caller hears the busy signal while the first caller has not been accepted yet (only possible for All).

- Configuring call strategy, priority and ringing delay for the existing groups.
 - Groups > Phone numbers > / > Basic Settings
- If necessary, configuring rework time for the existing groups.
 - Groups > Phone numbers > / > Reachability

Examples of calls illustrating the possible options are listed below. The following parameters are the same in all examples:

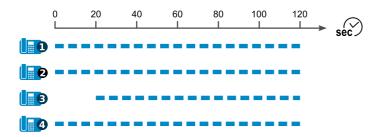
- · Number of phones logged in: 4
- · Configured time: 20 seconds
- Priority order: 10, 12, 13, 14
- Ringing delay for call strategy All: enabled for
- · CF delay time for CF on no reply: 20 seconds

AII

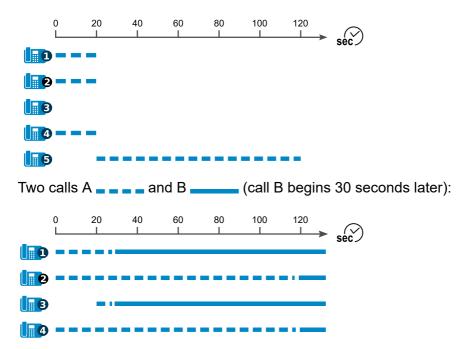
Like **All - Split Group**, but a second caller hears the busy signal while the first caller has not been accepted yet.

All - Split Group

One call:

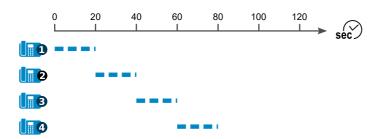


A call (call forwarding on no reply active on destination phone **[]**

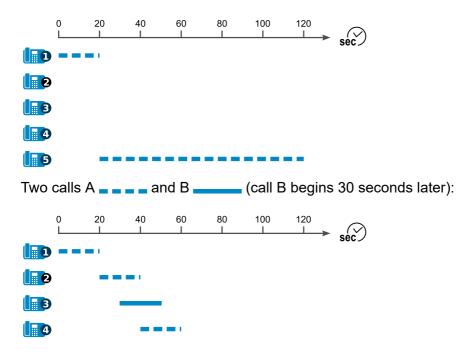


Linear - fixed - split group

One call:



A call (call forwarding on no reply active on destination phone **5**):

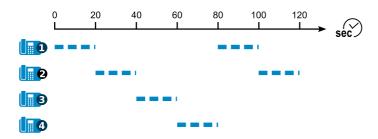


Linear - balancing - split group

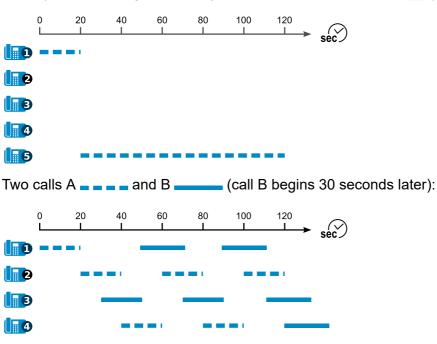
(This is different than **Linear - fixed - split group** due to the changing priorities only.)

Linear - fixed - rotating - split group

One call:



A call (call forwarding on no reply active on destination phone **5**):

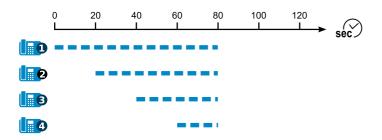


Linear - balancing - rotating - split group

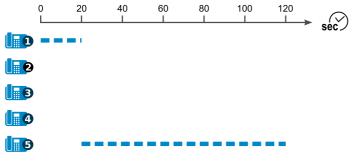
(This is different than **Linear - fixed - rotating - split group** due to the changing priorities only.)

Adding - fixed - Split Group

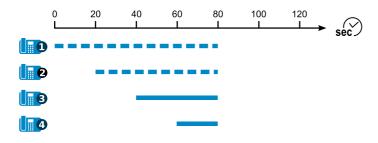
One call:



A call (call forwarding on no reply active on destination phone **[]**):



Two calls A _ _ _ and B _ _ (call B begins 30 seconds later):

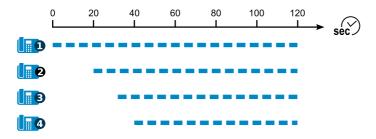


Adding - balancing - Split Group

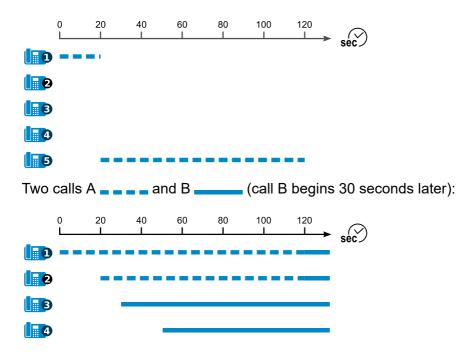
(This is different than **Adding - fixed - Split Group** due to the changing priorities only.)

Adding - fixed - rotating - Split Group

One call:



A call (call forwarding on no reply active on destination phone **5**):



Adding - balancing - rotating - Split Group

(This is different than **Adding - fixed - rotating - Split Group** due to the changing priorities only.)

Call Through

With Call Through, calls are passed through the PBX. This is used, when you do not want to reach a client via the private smartphone but via the company number. The smartphone calls the PBX and gets connected to the desired connection. Call Through allows the direct redial of any phone number.

This way, the called person does not receive the mobile number but the company number.

The phone number to be transferred is configured in the exchange line routing outgoing and is assigned to the according exchange line routing rule set of the Call Through function. Using the CLIP no screening function (availability on the account provided) allows a free assignment of the phone number to be transferred independently from the used connection.



Further help under Exchange line routing

Process

Outgoing calls from a mobile telephone are made as follows:

- 1. Dialling the predefined external number of the PBX for Call Through.
- The PBX accepts the call and signals this with a special tone. 2.
- 3. Entering the 6-digit Call Through PIN and *.
- 4. Based on the transmitted phone number and PIN, the PBX checks whether the caller has the authorisation for Call Through.
- 5. You will hear the dial tone and you have the following options:
 - You dial an internal telephone number.
 - You dial an external number with prefix.

- 6. After checking (block- and allowlist), the connection is established.
- Important: Emergency calls via this function are only possible under certain conditions.
 - Further help under <u>Emergency calls</u>
- Description of operation under <u>Call Through</u>

- If required, activating CLIP no screening on the account that is to be used for Call Through.
 - Exchange lines > Providers and Accounts > / >Miscellaneous
- Assigning a phone number of the account for Call Through.
 - Exchange lines > Exchange line routing > / > Exchange line routing incoming > Call distribution for
 - Note: If possible, have the mobile phone operator enter the PBX number as special number (higher cost savings).
- If required, creating an own exchange line rule set for call handling via Call Through.
 - Exchange lines > Exchange line routing > Exchange line routing outgoing
- Creating mobile phone numbers as contacts for verifying Call Through users.
 - User data > Contacts and assignment
- Configuring Call Through.
 - Functions > Call Through

Call waiting

If a call is already in progress, a call waiting signal can be used to indicate that another call partner is calling with the call waiting function

Process

1. During a conversation, a call waiting call is signalled.

You hear the call waiting tone. The call waiting caller hears the ringing tone.

- 2. You have the following options:
 - Reject the call waiting call.

The call waiting caller hears the busy signal.

- You take the call waiting call.

You are then connected with the call waiting call. The previous call partner is held.

- You disconnect the current call and accept the waiting call.

You are then connected with the call waiting call.

Description of operation under <u>Call waiting</u>

- Switching on call waiting separately for users.
 - User > Phone numbers > / > Reachability

Central caller list

The PBX can store up to 50 entries per user and the groups assigned to them in its central caller list. This function can be enabled separately for each user so they can access the saved data on their telephone. This is very useful if the roaming user function is also being used. When the user changes from using one telephone to using a different one, their calls are only stored in the central caller list. The local lists cannot access calls that have been made when the user was logged out or calls made on a different telephone.

A COMfortel D-series telephone is required to use the central caller list function. As soon as the user logs onto the telephone, the PBX provisions a key for calling the central caller list. The user can then use this key to display the entries, sorted by incoming calls, missed call or all calls. The local lists can still be accessed as usual.



Further help under Roaming User

Data

The central caller list of the PBX saves the following data of incoming and outgoing calls:

- · Name (last name, first name), if known in the central caller list
- Phone number
- Identity (for the user his local phone number, for groups the local group name)
- Date
- Time
- Call duration

If the function was previously or temporarily switched off for the user, the calls during this time of period are not saved in the list. This means they are not displayed on the phone when the function is switched on again.

If the phone is not registered during an incoming call or an update is in progress, the list in the phone is immediately updated when the connection is reestablished.

Functionality for the group

If the user is a member of a group, all calls in this group are saved in the list even when the function is disabled. They are displayed on the phone when the function is switched on again. This applies to all incoming and outgoing calls of all group members.

On an outgoing call of a user via a group, the group name is displayed and not the phone number of the user.

- Configuring central caller list separately for users.
 - User > Phone numbers > / > Settings

Certificates

Certificates ensure the identity of a server or client and encrypt data traffic.

The PBX can generate individual certificates itself, but also manage externally generated certificates, e.g. provider certificates.

Provider Certificate

A provider certificate must be imported into the certificate management and the selected in the provider's configuration.

Exception: If the provider supports Mutual SSL Authentication, the following data provided by the provider must be entered directly in the provider configuration:

- Client certificate
- Private key
- if necessary, any passphrase associated with the private key

Certificates for SIPS/SRTP internal

The following matching keys and certificates are required for SIPS/SRTP internal:

- · valid root certificate
- · valid PBX certificate
- · private key of the PBX certificate

The required certificates/keys can be generated in the PBX or externally generated and imported.

Some end devices require the PBX certificate or the Fingerprint of the root certifivate for verification. For this purpose, the PBX certificate can be exported and the Fingerprint copied or read.

Client certificates of the devices

Some devices have a client certificate. Thus, the identity of a user is bound to a unique digital certificate. This way, it can be determined who for example can access the user data.

Client certificates are stored in the PBX for the following devices:

- COMfortel D-series (all models)
- COMfortel 1400 IP / 2600 IP / 3600 IP
- COMfortel WS-500x
- · Snom IP phones
- · Yealink IP phones

If provisioning is to be enabled for devices without their own client certificate, access without a client certificate (insecure operation) must be enabled.

- · Storing certificates of the provider.
 - Administration > Certificates > Trustworthy
 - Exchange lines > Providers and Accounts > Name of the provider > / > SIP > SIPS
- Creating or importing certificates for SIPS/SRTP internal.
 - Administration > VoIP > SIPS/SRTP internal

Certificates

- · Managing certificates.
 - Administration > Certificates
- Switching access without client certificate on (unsafe operation).
 - Administration > Network > Server service > Provisioning

CLIP texts

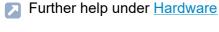
CLIP texts are displayed on VoIP phones and/or CLIP-capable analogue phones for specific calls.

- Enter CLIP texts for specific calls and functions.
 - Functions > CLIP texts

Conference rooms

This function enables a phone conference between several internal users. For this purpose, a conference room can be entered by calling an internal number.

A total of ten conference rooms can be configured. The number of conference rooms and the number of allowed users per conference room depends on the used hardware.



Process

- 1. The dial-in in a conference room is performed with the previously configured internal number.
- 2. The person dialling-in hears an announcement before entering the conference room.
- 3. The new participant is announced to others by a short tone.

Configuration

- Managing and assigning audio files for announcements.
 - Functions > Music on Hold/announcements
- · Creating and configuring conference rooms.
 - Internal Devices > Conference rooms

Alternative function on an internal phone:

- · Recording, listening to and deleting announcements.
 - Description of operation under <u>Conference rooms</u>

Configuration switchover

Some functions can, e.g. be configured differently according to the day, night, weekend, vacation or holidays. They are assigned to different configurations for this purpose.

Switching from one configuration to another can be carried out in two different ways:

- manually
- time-controlled (automatic configuration switchover)

Configuration-dependent functions

- Profile assignment for user (and with it e.g.: exchange line routing outgoing, call forwarding for users)
- Profile assignment for groups (and with it e.g.: announcement before answering, exchange line routing outgoing, call forwarding for groups)
- Profile assignment for voice mail and fax boxes (and with it e.g.: readiness)
- Exchange line routing incoming (call distribution, fallback, call forwarding for external phone numbers)
- Door terminal distribution
- · Relay settings in the configuration-dependent mode
- Further help under <u>Exchange line routing</u>
- Further help under Profiles

Configuration

- · Creating configurations.
 - Functions > Time control > Configurations
- Making settings configuration-dependent/time-controlled.
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Call distribution for
 - User > Profiles
 - Groups > Profiles
 - Voice mail and fax boxes > Profiles
 - External Devices > Door stations
 - External Devices > Relays (actuators)
- Switching on the authorisation for configuration switching for the involved internal phones.
 - User > Phone numbers > / > Authorisations
- Switching on/off automatic configuration switchover (time-controlled).
 - Functions > Time control > Configurations
- Creating switching times for automatic configuration switchover.
 - Functions > Time control > Switching times
- Switching on configuration.
 - Functions > Time control > Configurations

Alternative functions on an internal/external phone:

Configuration switchover

Configuration switchover

- Description of operation under <u>Configuration switchover</u>
- · Switching on/off automatic configuration switchover
 - Note: Using a phone, switch the system relay with operation mode automatic configuration switchover and the Identification number 900 (factory settings).
 - Description of operation under <u>Relays</u>

Contacts/LDAP

Contacts can be created in the PBX and managed in contact groups. When the LDAP server is activated, these contacts or contact groups can be transferred to phones connected to the PBX, provided they support it.

Contact Groups

The contact groups serve to manage contact data (phone numbers, names, contact pictures, addresses and e-mail / Internet addresses). For example, to organize large address books by department, the contact data can be assigned to different contact groups. Contact data can be made available to the connected telephones (if supported) via LDAP, separated according to contact groups.

The PBX distinguishes two types of contact groups:

Interface

Allows free assignment of contacts created manually, taken from the call data list or imported via vCard or CSV.

Google

Allows synchronisation with a Gmail account.

- Important: The synchronisation with a Gmail account is only available with feature level 2 or higher. If you want to use this function, please contact your dealer or system partner.
 - Further help under Feature level

New contacts are automatically assigned to the assigned default contact groups as soon as they are manually created, taken from the call data list or imported via vCard. However, contacts that have been created by

Contacts/LDAP

importing a contact file (csv import) are not automatically assigned to a standard contact group.

Note: If a large number of phones are connected to the PBX, or large volumes of contact details are to be uploaded, it may in certain cases take several hours before the contact data is available to all the phones after the PBX has been commissioned.

LDAP server

The LDAP server is needed to transfer the PBX's contact data to those IP end devices which support this transfer method (e.g. SIP phones with LDAP support). Any contact data that is input directly on a phone is not transferred to the central contact files.

The LDAP server's IP address matches the PBX's IP address and is predefined.

When switched off, the security and performance of the PBX are improved. No contact data can be transferred to the IP end devices.

Auerswald/FONtevo provides LDAP support for the following IP end devices:

IP telephone	COMfortel 1200 IP
IP telephones	COMfortel 1400 IP / 2600 IP / 3600 IP (from firmware version 2.4A)
IP telephones	COMfortel D-100/D-110/D-200/D-210/D-400/D-600
IP DECT phones	COMfortel M-5x0/M-7x0 via IP DECT base stations COMfortel WS-500S and COMfortel WS-500M

Note: Data is sent to D-series telephones via LDAPS, in encrypted format.

Enter the following values into the IP end device to transfer the contact data from the LDAP server of the PBX to the IP end device:

	Selected contact groups (e.g. in the factory settings)
User name	cn=auerswaldschandelah,dc=auerswald
BaseDN	dc=auerswald
Password	automatically generated password
	Contacts for individual users (e.g. for the internal number 31)
User name	
User name BaseDN	internal number 31)

- Note: You must enter the complete character string. Ensure you type the capital and lower-case letters correctly, and do not type any blank spaces before or after the , and =space.
- Note: In order to avoid input errors, on the web interface in the lines below the entry and list fields you can mark and copy the user name and BaseDN.
- For more information on data exchange with the LDAP server, take a look at the Privacy Policy of the COMtrexx. You will find this in

the Internet under: <u>Auerswald Support/FONtevo Support</u> or here: **Dashboard > Status information > Legal > Data protection**

- Creating and assigning contacts.
 - User data > Contacts and assignment > Contacts and assignment > + New
- Creating contact groups.
 - User data > Contacts and assignment > Contact Groups > + New
- Defining authorisations for contact groups.
 - User data > Contacts and assignment > Authorisations
- Switching on the default standard contact group for the user.
 - User data > Contacts and assignment > Authorisations
- Activating LDAP server.
 - User data > Contacts and assignment > Settings

Date and Time

The PBX requires a system time. The available time information is a prerequisite for functions such as automatic configuration switching but also encrypted telephony.

The PBX has the following features for this purpose

- an internal clock
- a buffered real-time clock that stores the time for approx. two hours in the case of a power outage (only COMtrexx Business)

The PBX time can be updated as follows:

- Set time manually
- Set time by taking over from the PC
- Set time automatically by querying the entered NTP servers

The associated telephones/devices usually also require mandatory time. For this purpose, the PBX provides one of the following NTP servers in the devices:

- PBX as NTP server (devices in the same network are then supplied with the system time)
- · entered NTP server
- Default NTP pool
- Important: If the units are neither in the network of the PBX nor are they able to reach addresses on the Internet, an individually created provisioning file is required in which an accessible NTP server is entered.
- For more information on data exchange with public NTP servers, take a look at the Privacy Policy of the COMtrexx. You will find this

in the Internet under: <u>Auerswald Support/FONtevo Support</u> or here: **Dashboard > Status information > Legal > Data protection**

- Configuring PBX time.
 - Administration > Date and Time
- · Providing the PBX as NTP server.
 - Administration > Date and Time > Automatic distribution

DECT system

This PBX supports the operation of the IP DECT base stations COMfortel WS-500S and COMfortel WS-500M. The following handsets can be operated in connection with these devices as IP DECT phones on the PBX:

- COMfortel M-510
- COMfortel M-520
- COMfortel M-530
- COMfortel M-710
- COMfortel M-720
- COMfortel M-730

Singel-cell system/Mini multi-cell system with the base station COMfortel WS-500S

You can use a COMfortel WS-500S to create a single-cell system with a base station.

The device has an integrated DECT Manager with Integrator. With this, the device manages the handsets via its own web configurator and provides the DECT functions. For up to 20 handsets, the device provides communication between the handset and the PBX.

A Mini multi-cell system can be built up from up to 3 COMfortel WS-500S. Here, a device acts as the central management and configuration station of this system and takes on various device roles.

When delivered, the COMfortel WS-500S is configured as a "base station with DECT manager and integrator". To set up the mini multi-cell system, the other units must be configured as "Base station only".

With a licence upgrade, the unit can be integrated into larger multi-cell systems.



Further information can be found in the Instructions of the base station.

Multi-cell system with the base station COMfortel WS-500M

One or more COMfortel WS-500Ms can be used to set up a multi-cell system with one or more base stations and a central management and configuration station. In this case, the COMfortel WS-500M performs various device roles. Different extension levels are possible here.

With the "Integrator" device role, the device manages the handsets via its own Web Configurator.

With the "DECT Manger" device role, it manages the synchronisation of the base stations and ensures the communication from the PBX to the respective base stations.

When delivered, the COMfortel WS-500M is configured as a "Base station only". To set up the multi-cell system, one device must be configured as a **DECT** Manager and Integrator.

Depending on the expansion stage of the DECT system, up to 112 handsets (limited by the maximum number of users of the PBX) can be connected.



Further information can be found in the Instructions of the base station.

General information about the configuration

Connection and operation of the IP DECT base stations COMfortel WS-500S and COMfortel WS-500M to the PBX is via the Ethernet port and correspondingly assigned VoIP channels.

Depending on the procedure (e.g. use of provisioning or manual configuration), the base stations and handsets are configured and maintained on the web interface of the PBX and in the Web Configurator of the base station. See the descriptions below.

Only those base stations are set up in the PBX for which the device role "DECT Manager and Integrator" is activated. Base stations with the role "Base station" are not set up in the PBX, but for multi-cell operation in a previously determined base station with the role "DECT Manager and Integrator".

The handsets are operated in connection with the base station as IP DECT phones on the PBX.

Via provisioning, all handsets are supplied with the contacts of the PBX if LDAP is set up and activated.

Requirements for first commissioning

For information e.g. on planning the DECT network, setting up the base station in the local area network (LAN) and setting the device role, see the operating instructions of the base station.

First commissioning via provisioning

- Important: The base station to be set up may only be connected to the local network after the provisioning file has been assigned. Provisioning cannot be initiated automatically if the base station is connected before then.
- Important: Base stations with the device role "Base station" are not configured in the PBX but on a base station with the role "DECT Manager and Intergrator" (see Advanced Information of the base station).

- 1. If required, create an individual provisioning file for the device.
 - Telephones > Provisioning > New
- 2. Set up users with the phone numbers of the handsets in the PBX.
 - User > Phone numbers > +
- 3. Set up the COMfortel WS-500S/COMfortel WS-500M base station in the PBX.
 - Telephones > Assignment > New
 - Type: DECT base
 - MAC address: sticker on the back of the device
 - Provisioning file: default file is automatically assigned; the individually created file must be assigned manually
- 4. Set up COMfortel M-5x0/M-7x0 handsets in the PBX and assign them to a base station.
 - Telephones > Assignment > New
 - Type: DECT handset
 - Device IPUI: on the packaging label; alternatively in the handset
 Control key + *#06#
 - COMfortel WS-500x base station: Select the base station
- 5. Assign the handsets to users.
 - Telephones > Assignment> list field User
- 6. If required, set up a multi-cell system (with more than one base).
 - For the leading base station: device roll**DECT Manager and** Integratorset up directly on the device
 - For a Mini multi-cell system with more than one base station, the additional base stations require manual setup in the base

station web configurator (see Advanced Information of the vbase station).

- 7. Connect the leading base station to the network via LAN cable and wait for the boot process.
 - The base station restarts and the provisioning file is automatically transmitted from the PBX to the base station and the device is configured.
- 8. After successful provisioning, the base station remains in login mode for one hour. Meanwhile, register the handsets with the base station.
 - On the handset: Settings > Logging in > Registering a handset

Manual first commissioning

- Important: Base stations with the device role "Base station" are not configured in the PBX but on a base station with the role "DECT Manager and Intergrator" (see Advanced Information of the base station).
- 1. Set up users with the phone numbers of the handsets in the PBX.
 - User > Phone numbers > +
- 2. Set up the COMfortel WS-500S/COMfortel WS-500M base station in the PBX
 - Telephones > Assignment > New
 - Type: DECT base
 - MAC address: sticker on the back of the device
 - Provisioning file: assigning no file with ---
- 3. Set up COMfortel M-5x0/M-7x0 handsets in the PBX and assign them to a base station.

- Telephones > Assignment > New
- Type: DECT handset
- Device IPUI: on the packaging label; alternatively in the handset
 Control key + *#06#
- COMfortel WS-500 base station: Select the base station
- 4. Assign the handsets to users.
 - Telephones > Assignment> list field User
- 5. Connect the leading base station to the network via LAN cable and wait for the boot process.
- Log on to the web configurator of the base station and configure the PBX, the base station and the handsets (see operating instructions for the base station under **Initial Commissioning**).

Checking for firmware update

After the configuration of the base station, it should be checked via the Auerswald homepage whether a firmware update is available for the device.

Dialled calls

For outgoing calls, it is distinguished between internal and external calls based on the dialled number.

Further help under **Exchange line request**

Outgoing external calls are controlled by the rule set of the exchange line routing outgoing.

Further help under Exchange line routing

When dialling a number without area code, the PBX determines the connection to be made based on the location.

- Further help under Locations
- Description of operation under <u>Dialled calls</u>

Do-not-Disturb

Do-not-Disturb offers the option of blocking internal and external calls as well as blocking group calls on a specific phone.

Do-not-Disturb has no effect on dialled calls and callbacks.

With the aid of block- and allowlists incoming, it is possible to block or allow calls of single phone numbers.

Further help under <u>Block- and allowlist</u>

Process

Do-not-Disturb proceeds as follows:

- 1. A caller hears the busy tone.
- 2. An internal caller can reach a phone with call protection in an emergency using a priority call.
- Description of operation under <u>Unsuccessful outgoing calls</u>

Configuration

- Switching on Do-not-Disturb separately for users.
 - User > Phone numbers > / > Reachability

Alternative functions on an internal phone

- Switching on Do not disturb mode
- Description of operation under <u>Do-not-Disturb</u>

Door functions

The PBX supports the connection and control of various types of door terminals. In addition to a door bell, ringing at the door can in this way also be indicated by internal or external phones or groups. Each door bell button can be assigned an individual target number. You can also carry out various door functions such as speech connection with the door and actuation of the door opener from internal phones.



Description of operation under Door functions

- · Creating a door terminal.
 - External Devices > Device wizard
- · Configuring door terminals.
 - External Devices > Door stations > /
- Creating a door call and assigning it to the door terminal.
 - External Devices > Door stations > +
- · Configuring door calls.
 - External Devices > Door stations > /
- Creating a relais and assigning it to the door terminal.
 - External Devices > Door stations > +
- Switching on the authorisation for opening the door separately for the involved internal phones.
 - User > Phone numbers > / > Authorisations

Door functions

- Switching on the authorisation for opening the door configuration-dependent/time-controlled for the involved internal phones.
 - User > Profiles > / > Authorisations

E-mail

The PBX supports the following e-mail functions:

- · System e-mails
- Forwarding of voice mail and fax messages

The settings described here must be made centrally in the PBX and can then be freely assigned to the individual e-mail functions of the PBX.

You can configure up to four outgoing mail servers in the PBX. These can be freely assigned to individual e-mail functions of the PBX (system e-mails, forwarding of voice mail and fax messages).

An outgoing mail server is responsible for the transfer of e-mails. You will receive the access data for the outgoing mail server to be used from your Internet service provider, for example.

The PBX supports any number of logos (limited by the total memory). These can be freely assigned to individual e-mail functions of the PBX (system e-mails, forwarding of voice mail and fax messages).

For the e-mails to be sent by the PBX, several language templates are included in the scope of supply. The PBX supports additional user-defined language templates which can be generated on the basis of the existing language templates.

These language templates include different text elements (boilerplates) which are used to create the e-mails. The language templates are in *.xml format.

- · Configuring outgoing mail server.
 - Administration > E-mail > Outgoing mail server

E-mail

- · Managing logos.
 - Administration > E-mail > Logo management
- Managing language templates.
 - Administration > E-mail > Language Templates
- Assigning an outgoing mail server, logo and language template to a voice mail/fax box.
 - Voice mail and fax boxes > Phone number > / > Box settings
- Assigning an outgoing mail server, logo and language templates to the system e-mails.
 - Administration > Maintenance > Message Configuration

Emergency calls

In this case, calls from an internal telephone to configured emergency numbers are regarded as emergency calls. Special rights apply to calls to emergency numbers. The emergency numbers available at a location must be configured in the PBX for this purpose.

The exchange line access number required for other telephone calls (factory settings) can be omitted for emergency calls. This has the benefit that even people who do not know how to use exchange line access numbers can make emergency calls. If the emergency priority circuit is enabled (factory settings), emergency calls take priority over other calls. If required, a call can be interrupted to enable an emergency call.

Special numbers such as 112 are dialled correctly, without a prefix, without colliding with internal phone numbers.

When a call is made to an official emergency number, the provider transmits the location to the emergency call centre. Localization is necessary if the caller is no longer able to give his name and address. The provider usually uses the contractor's address or the address of the requested location.



▲ Caution: It is essential that the address the provider has stored for your accounts matches the location of your PBX.

Accounts whose location does not match your PBX's location cannot be used for emergency calls. You must set up and specify the correct location for these accounts.



▲ Caution: When telephones are being operated in several locations (e.g. used by several company offices, external staff or home offices), remember that emergency calls to these telephones can only be made if specific requirements have been met.

In addition, the providers and accounts can be prevented from making emergency calls if you disable the emergency call ability. Make this setting for providers and accounts that are unreliable or that should be prevented from making emergency calls for other reasons.

If a telephone has to be blocked, to prevent it being used for emergency calls, for example, to prevent misuse, this block can be set for the associated user.

Softphones (also called soft clients) are very often not permanently bound to a location and are therefore created without a location by default. An emergency call with the softphone via the PBX is therefore not possible.

Caution: Label the telephones that cannot be used for emergency calls. Inform all the users affected by this that they must use a different telephone that is available at that location to make emergency calls. For an emergency call via a smartphone, do not use a softphone installed on it, but normal mobile telephony.

Further help under Locations

Requirements for emergency calls at several locations

Each official emergency number is assigned to a location and, therefore, to a country. Emergency calls to official emergency numbers are only made using accounts whose location is the same as the location of the telephone from which the call is made. This ensures users can access the emergency services in their country and that the provider sends the correct location to the emergency call centre.

The ability for remote locations such as other company offices in the same country and abroad to make emergency calls can be enabled by assigning each of them their own valid account.

If another company office were located in the same local telephone network as the PBX, e.g. in the same district or in another street, it could

use the same emergency numbers. However, in this situation, the provider would not be able to send the correct address to the emergency call centre. In the worst case, this might mean that an ambulance was sent to the wrong location. This problem can be resolved by the provider a separate account to the address of the second location. In the PBX, another location, with a different name, must then be set up for this account and the telephones associated with it.

▲ Caution: If a telephone's ability to make an emergency call cannot be guaranteed (e.g. in the case of telephones used in a home office), mark it accordingly. Inform all the users affected by this that they must use a smartphone or a different telephone that is available at that location to make emergency calls.

Legal emergency numbers

Legal emergency numbers here are special numbers that do not follow the standard numbering scheme, therefore, do not start with a country or local area code. Each phone number is assigned to a location and therefore, to a country. For calls to legal emergency numbers, the used account must be assigned to the location of the used telephone.

The standard depends on the configured countries under locations. Entries remain but can be manually deleted when the location settings are changed.

User-defined emergency numbers

User-defined emergency numbers here are numbers that follow the standard numbering scheme. Each phone number is assigned to a location and therefore, to a country. For calls to user-defined emergency numbers, the according standard account is used.

Service numbers

In addition, local service numbers, such as numbers for doctors or local authorities can be set up in the PBX. No special rights are applied on calls to special numbers. Service numbers here are special numbers that do not follow the standard numbering scheme, therefore, do not start with a country or local area code. Each service number is assigned to a location and therefore to a country. For calls to service numbers, the used account must be assigned to the location of the used telephone.

Process

An emergency call is made like this:

1. The user dials an emergency number on an internal telephone.

The following rules/restrictions apply:

- Depending on the configuration, emergency calls can be made either with or without dialling the exchange line access number.
 This has the benefit that even people who do not know how to use exchange line access numbers can make emergency calls.
- 2. The PBX system compares the phone number that was dialled with the emergency numbers set up in the system.

The following rules/restrictions apply:

- Official emergency numbers are country-specific, and therefore also location-specific. The validity of these numbers is always defined by the location of the telephone from which the call is made.
- 3. If the emergency number is recognised, the PBX checks whether the emergency call can be made.

The following rules/restrictions apply:

- An emergency call cannot be made if the associated user is prevented from making emergency calls.
- An emergency call cannot be made if the emergency number dialled is on a valid blocklist.
- The PBX checks whether suitable accounts are available for the emergency call.

The following rules/restrictions apply:

- Emergency calls to official emergency numbers are only made using providers/accounts whose emergency call ability is enabled.
- Emergency calls to official emergency numbers are only made using accounts whose location is the same as the location of the telephone from which the call is made.
- If the phone cannot make an emergency call, the user is informed by an announcement.
- 5. If suitable accounts are found, the PBX makes the emergency call.

The following rules/restrictions apply:

- If the account that is preferred in exchange line routing is available and present, the emergency call is made using this account when possible.
- Emergency calls can be configured to take priority over other calls.
 If required, a call can be interrupted to enable an emergency call.
- The phone number is always sent along with an emergency call.
 If number presentation suppression is switched on, it is disabled automatically when the emergency call is made.

- Location where the PBXs and individual accounts are operated.
 - Exchange lines > Exchange line request/locations
- Making general settings for emergency calls and configuring emergency numbers.
 - Exchange lines > Emergency calls
 - **Note:** Legal emergency numbers are preset with the location setting.
- Enabling/Disabling emergency call ability for providers.
 - Exchange lines > Providers and Accounts > Name of the provider > / > Settings
- Enabling/Disabling emergency call ability and configuring locations differing from the standard for accounts.
 - Exchange lines > Providers and Accounts > Name of the account > / > Settings > Network Provider
- Configuring emergency call routing.
 - Exchange lines > Exchange line routing > Exchange line routing outgoing
- Configuring emergency call reaction for users.
 - User > Phone numbers > / > Exchange line settings
- Managing and assigning audio files for emergency call announcements.
 - Functions > Music on Hold/announcements > File management

- Functions > Music on Hold/announcements > File assignment
 - Note: The PBX is supplied with emergency call announcements in English and German. You can create other emergency call announcements in the local language at a later point in time.

Alternative function on an internal phone:

- · Recording, listening to and deleting announcements.
 - Description of operation under <u>Emergency Call Announcement</u>

Exchange line authorisation

The exchange line authorisation determines which phone number types can be dialled (e.g. phone numbers in the local phone network). This is determined in the exchange line routing.



Further help under **Exchange line routing**

Exchange line request

Exchange line request describes the process of an exchange line allocation.

The PBX distinguishes between two types of exchange line requests.

Automatic exchange line request

In factory settings, the automatic exchange line request is switched on.

Neither for internal nor for external numbers a prefix is required (exception: ** for targeted VoIP access). The PBX automatically distinguishes between internal and external numbers with the adjustment of the dialled phone number and the internal dial plan.

- If the phone number is contained in the dial plan, the PBX establishes an internal connection
- If the phone number is not contained in the dial plan, the PBX establishes an external connection.

The following rules/restrictions apply:

- In order to have a correct distinction between internal numbers and phone numbers in the local area network, the internal dial plan must not contain phone numbers that also occur within the local area network. To avoid conflicts, phone numbers of the local area network should always be dialled with local area code.
- Also when the entered phone number corresponds with the internal dial plan, the PBX waits up to 4 seconds for whether additional digits for an external number are being entered, before it dials the internal number. To avoid waiting time, you can subsequently enter #. The PBX therefore recognises the input as finished and instantly starts dialling.

Further help under <u>Internal numbers</u>

Internal phone

The user has to dial an external number with preceding exchange line access number.

The exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Changing the exchange line access number reduces the dial plan.

Further help under Internal numbers

Targeted VoIP access point

Each account can be assigned its own exchange line access number for targeted dialling via a specific account. Also this exchange line access number reduces the dial plan.

The check for emergency calls is always performed on the basis of where the telephone making the call is located. For this reason, the account is only used as the preferred account.

- Further help under Internal numbers
- Description of operation under <u>Targeted VoIP access point</u>

- · Configuring exchange line request and, if necessary, exchange line number
 - Exchange lines > Exchange line request/locations
- Assigning a separate exchange line access number to an account.

- Exchange lines > Providers and Accounts > / > Access data

Exchange line routing

The exchange line routing adjusts the routing of incoming external calls (exchange line routing incoming) and of outgoing external calls (exchange line routing outgoing).

Exchange line routing incoming

The exchange line routing incoming adjusts the call distribution. The call distribution defines the internal numbers to which incoming external calls are signalled, e.g. which internal phones ring when you receive a call.

For the number block of a system connection, linear call distribution always applies during initial setup. This means that external calls to individual DDIs are signalled to identical internal numbers. By adjusting the first internal destination, the internal destinations can be relocated into another area of the internal dial plan. For example: calls over the numbers 920010-920099 are signalled to the internal numbers 10-99. By adjusting the first destination to 210, the calls are signalled to the internal numbers 210-299.

Fallback destinations can be configured for non-existing internal destinations

A number block can be separated as required by inserting a new number block for a partial area. This allows to separate the available phone numbers into differently treated areas. For example calls over the phone numbers 920010-920099 can be distributed to the internal number block 210-239 and 540-599. A new number block inserted for this purpose is always created with linear call distribution. The remaining number blocks keep their settings. If a number block is deleted, you will be asked which number block should be connected to. The settings of the chosen number block are adopted and any existing call forwardings are deleted.

Further help under Internal numbers

In addition, external calls can here be directly forwarded to an external phone number.

Further help under <u>Call Forwarding for external numbers</u>

Also individual external phone numbers are assigned to the Call Through function and remote programming.

- Further help under Call Through
- Further help under <u>External access</u>

Also block- and allowlists incoming are created here. These lists can be used to define individually for users and groups which callers are blocked (e.g. anonymous callers) and which callers are released despite Do-not-Disturb.

Further help under Block- and allowlist

Exchange line routing outgoing

The exchange line routing adjusts how outgoing external calls are handled depending on the dialled number. For this purpose, there are several routing rules summarized in one rule set. These rule sets are created to be used system-wide and are freely assigned to users, groups or special functions

A routing rule defines how an individual phone number or a whole number range or phone number category is handled. The following phone number categories are part of each rule set: **Emergency calls**, **International**, **National** and **City**.

A routing rule defines which one of the available accounts (preferred) is used for initiating an outgoing external call. By selecting an individual

Exchange line routing

account a preferred exchange line can be realised. Instead of calling an account, busy can be output and with this the exchange line authorisation be restricted.

In addition, a routing rule defines which phone number is presented. Using the CLIP no screening function (availability on the account provided) allows a free assignment of the phone number to be transferred independently from the used connection.

Further help under Number presentation

For maximum availability, the routing rule for emergency calls can only be configured to a limited extent.

Further help under <u>Emergency calls</u>

Also block- and allowlists outgoing are created here. These lists are assigned to the exchange line routing rule sets. Blocklists outgoing define which external phone number can be called. Allowlists outgoing define which external phone numbers differing from the routing rules can be called.

Further help under Block- and allowlist

- · Setting call distribution.
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Call distribution for
- · Creating new exchange line routing rule set.
 - Exchange lines > Exchange line routing > Exchange line routing outgoing > + New
- · Configuring exchange line routing rule set.

- Exchange lines > Exchange line routing > Exchange line routing outgoing > /
- · Assigning exchange line routing rule set to a user.
 - User > Phone numbers > / > Exchange line settings
- Assigning exchange line routing rule set to a user configurationdependent/time-controlled.
 - User > Profiles > / > Exchange line settings
- Assigning exchange line routing rule set to a group.
 - Groups > Phone numbers > / > Exchange line settings
- Assigning exchange line routing rule set to a group configurationdependent/time-controlled.
 - Groups > Profiles > / > Exchange line settings
- Assigning exchange line routing rule set to the Call Through function.
 - Functions > Call Through > Exchange line settings

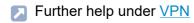
External access

The PBX is usually configured via an internal network connection. The PC you use to do this must be connected to the same network as the PBX.

Alternatively, the PBX can be configured remotely via an Internet connection (DSL).



▲ Caution: Every time port forwarding is performed, there is a security risk. Therefore, only carry out remote configuration via a VPN tunnel.



Different functions such as configuration switching can, if necessary, also be proceeded using an external phone.

COMtrexx Control Center (CCC)

The COMtrexx Control Center (hereinafter called CCC) serves as a functional extension of the PBX for resellers and system administrators. It provides remote management, control and monitoring of services and devices.

- For more information on CCC, take a look at the manual of the CCC.
- For more information on data exchange with the CCC, take a look at the Privacy Policy of the CCC.
- Important: To create remote backups, this function must first be set up in the PBX.
 - Further help under **Backup**

Configuration

Activate the proxy service to use the CCC.

- Administration > VoIP > VoIP configuration > Proxy server for COMfortel SoftPhone > Use proxy
- After request by the CCC, start the requested pairing.
 - Administration > Service > External access > Access for CCC
 Confirm CCC access
- Allow monitoring via the CCC.
 - Administration > Service > External access > Access for CCCMonitoring

Switching functions on/off via an external phone

Requirements:

- Remote switching number which is configured in the PBX
 - Exchange lines > Exchange line routing > Exchange line routing incoming
- DTMF-enabled telephone or DTMF tone generator
- Knowledge of the PIN and digit sequence required for the function
- 1. Call the remote switching number.

The PBX automatically accepts the call.

You will hear the external confirmation tone (1 second).

2. Enter with DTMF the required digit sequence.

You will hear the external confirmation tone again.

Possible functions on an external phone:

External access

- Switching on/off the calling of the user's mobile telephones (FMC)
 - Description of operation under <u>FMC</u>
- · Switching Follow-me on/off
 - Description of operation under Follow-me
- · Switching the configuration
 - Description of operation under <u>Configuration switchover</u>
- · Switching relays on/off
 - Description of operation under Relays

Fax

An analogue fax machine can be connected and configured on a COMtrexx Business with a 4FXS module.

As alternative to an analogue fax machine, there is the possibility to create one or more fax boxes that can be assigned to the users and groups.

Further help under Voice mail and fax boxes

- Configuring an analogue fax as a device with internal number.
 - External Devices > Fax
- · Configuring call distribution for fax machines.
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Call distribution for

Feature level

You can find out the feature level of your PBX here:

Dashboard > Feature level

Some functions depend on the feature level of the PBX. If there is a difference between the current feature level of the PBX and the maximum possible feature level, the full range of functions of the PBX cannot be used

For an upgrade of the feature level and the associated use of additional functions, please contact your specialist dealer or system partner.

If the PBX has the maximum feature level and not all functions are available, check the firmware version and, if necessary, carry out a firmware update.

Further help under Firmware Update

You will find out which functions you can use with which feature level and which firmware version in the lists below.

If the PBX does not yet have the latest firmware, you will find information on the currently available feature levels under Feature level.

Feature level 2 - Version 1.8

OpenVPN Server

In order to be able to securely connect devices to the PBX, the PBX offers an integrated OpenVPN server.

Further help under VPN

Google contacts

The contacts of the PBX can be synchronised with existing Gmail accounts.

Further help under Contacts/LDAP

Call flow display

For incoming calls, a COMfortel D-series telephone displays which destination was called (external call number, group, forwarded destination).

Further help under Incoming Calls

Feature level 2 - Version 1.10

Remote Backup

A backup can be stored password-protected on an external server and also restored from it.

Further help under Backup

Firmware Update

When firmware is updated, errors are corrected or new features are installed. To do this, the device establishes a connection to a server in order to download the most current firmware (the device settings remain intact).

- Note: If you have just performed a firmware update and an error message appears when you log into the web interface, please empty your web browser's cache or wait for about five minutes before logging in again.
- Note: Some of the functional upgrades provided as part of a firmware update require additional configuration parameters. Sometimes not all these parameters can be determined automatically from the previous firmware version. This is why you need to edit some parts of the configuration manually in this type of update.
- **Note:** After a firmware update, you usually require a new version of the instructions.
- You will find the up-to-date instructions on the Internet under <u>Auerswald Support/FONtevo Support</u>.

When an instant firmware update is activated via the update server, the firmware update is initiated automatically. A connection to the update server is set up and then the server sends the latest firmware to the PBX.

- **Important:** For the update service, the PBX requires access to the update server. Make sure that the address and port are reachable.
 - Further help under Network
- For more information on data exchange with the update server, take a look at the Privacy Policy of the COMtrexx. You will find this in

the Internet under: <u>Auerswald Support/FONtevo Support</u> or here:

Dashboard > Status information > Legal > Data protection

Other requirements for firmware update via update server

Requirements for firmware update via update server and for the search of firmware updates:

- PBX connected to broadband Internet access
- Configured DNS server and, if necessary, an HTTP proxy server
- · Configured gateway
 - Further help under Network

Configuration

Firmware update options:

- Manual Firmware Update
 - Administration > Maintenance > Update and Restart > Store firmware in the PBX
- · Firmware update via update server
 - Administration > Maintenance > Update and Restart > Firmware update via update server
- · Restoring firmware
 - Administration > Maintenance > Update and Restart > Restart

Alternative function by pressing a key on the PBX (only COMtrexx Business)

- · Restoring firmware
 - Description of operation under <u>Button</u>

Firmware Update

Alternative functions on an internal/external phone

- Initiating an automatic PBX update
 - Description of operation under <u>Firmware Update</u>

FMC

Use FMC (abbreviation for Fixed Mobile Convergence) to set up a mobile telephone as the user's telephone, so it can be accessed at any time on the mobile phone network.

Further help under <u>User</u>

At the same time, the mobile telephone configured for FMC is also given the authorisation to handle outgoing calls via the PBX. On outgoing calls, FMC acts like Call Through except that FMC is user-specific. The called person does not receive the mobile number or the central company number but the number that is also transferred from the office phone of the user.

Further help under <u>Call Through</u>

The phone number to be transferred is configured in the exchange line routing outgoing and the according exchange line routing rule set is assigned to the user. Using the CLIP no screening function (availability on the account provided) allows a free assignment of the phone number to be transferred independently from the used connection.

Further help under Exchange line routing

An FMC call is displayed in the call data.

Further help under Call data

Process

Outgoing calls from a mobile telephone are made as follows:

- 1. Dialling the predefined external number of the PBX for Call Through.
- 2. The PBX accepts the call and signals this with a special tone.

- 3. Entering the 6-digit user PIN and *.
- 4. Based on the transmitted phone number and PIN, the PBX checks whether the caller has the authorisation for FMC.
- 5. You will hear the dial tone and you have the following options:
 - You dial an internal telephone number.
 - You dial an external number with prefix.
- 6. After checking (block- and allowlist), the connection is established.
- Important: Emergency calls via this function are only possible under certain conditions.
 - Further help under <u>Emergency calls</u>
- Description of operation under <u>FMC</u>

- If required, enable CLIP no screening on the account that is to be used for FMC/Call Through.
 - Exchange lines > Providers and Accounts > / > Miscellaneous
- Assigning a phone number of the account for Call Through.
 - Exchange lines > Exchange line routing > / > Exchange line routing incoming > Call distribution for
 - Note: If possible, have the mobile phone operator enter the PBX number as special number (higher cost savings).
- If required, create your own exchange line routing rule set to specify how the user's calls are handled.

- Exchange lines > Exchange line routing > Exchange line routing outgoing
- Selecting exchange line routing separately for users.
 - User > Phone numbers > / > Exchange line settings
- Selecting exchange line routing outgoing configuration-dependent/ time-controlled for users.
 - User > Profiles > / > Exchange line settings
- Setting up the maximum duration for external calls and reaction to emergency calls.
 - Functions > Call Through
- Setting up the mobile telephone as a telephone and assigning it to a user.
 - Telephones > Assignment

Alternative functions on an internal/external phone:

- Switching on/off the calling of the user's mobile telephones
 - Description of operation under <u>FMC</u>

Follow-me

Follow-me is like a call forwarding unconditional for users or groups that can be configured on any phone (instead of on the forwarding phone). This gives you the option of moving from room to room and taking the calls with you. External destination numbers can also be configured.

For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply for the user are proved before a Follow-me is executed.

Further help under **Exchange line routing**

Follow-me can be configured from different phones:

Configure Follow-me to an internal destination phone: With the help of this function you can go from room to room and take your calls with you. You can configure the call forwarding of your calls from the phone (destination phone) where you currently are. But you can also take the calls of several users/groups at your phone.

Configure Follow-me from any available internal phone: If you have already left your desk, you can forward your calls to another phone. This way, it is possible to forward the calls to your mobile phone or to an internal substitute.

Configure Follow-me from external: If you have already left the house, you can forward your calls to another phone. This way, it is possible to forward the calls to your mobile phone or to an internal substitute.

Note: The PIN of the forwarding user is required for configuration on an external phone.

Configuration

- Enabling the following authorisations separately for the involved internal phones: CF (groups) / Follow-me, Controlling of the PBX via phone, Configuring CF (sub) to ext. number.
 - User > Phone numbers > / > Authorisations
- Enabling the authorisations for the involved internal phones configuration-dependent/time-controlled:
 - User > Profiles > / > Authorisations
- Configuring Follow-me for users/groups.
 - User > Phone numbers > / > Reachability
 - Groups > Phone numbers > / > Reachability

Alternative functions on an internal/external phone:

- Enabling/disabling Follow-me to an internal destination phone
- Enabling/disabling Follow-me on any available internal phone
- Enabling/disabling Follow-me on an external phone
 - Description of operation under Follow-me

Groups

Phone numbers and specific attributes can be assigned to a group in the same way as to a user.

Membership in a group does not necessarily mean that a user receives calls sent to the group. A user that is logged out takes on no group functions. This can be especially important for employees of a support department, who should not be available to customers around the clock. but instead rotate in and out of the hotline.

The Rework time attribute can be used to specify that group members cannot be called after they have received an internal or external call (for example, so that additional work can be performed on the call). During this time, the Do-not-Disturb function is enabled for the user.

Different call strategies can be used to assign inbound calls to group members "properly".



Further help under Call Strategy

There are three different ways of logging into a group:

Only incoming

When logging into a group as **Incoming**, the user then finds himself in the call distribution of the group in question for internal, exchange line and door calls.



Description of operation under Groups

Only outgoing

When logging into a group as **Outgoing**, the user acquires a number of characteristics/access rights from the group which replace his own

Groups

characteristics/access rights as an individual user for outgoing calls. A user can only be logged on in a single group as **Outgoing** even if he is a member of more than one group.

- Note: If a user logs into different groups as Outgoing, he will only remain logged in as Outgoing in the group that he logged into last.
- Description of operation under Groups

Incoming and outgoing

When logging into a group as **incoming and outgoing**, the user then finds himself in the call distribution of the group in question for internal, exchange line and door calls and also acquires a number of additional characteristics/access rights from the group which replace his own characteristics/permissions as an individual user for outgoing calls.

- Note: If a user logs into different groups as incoming and outgoing, he will only remain logged in as incoming and outgoing in the group that he logged into last. And in all other groups, he is then only incoming.
- Description of operation under <u>Groups</u>

Main and subgroup

If a group (subgroup) has been configured in the PBX as member of a group (main group) and a user is at the same time member of these two groups, he will only be considered as a member of the main group when a call comes in.

Subgroups (e.g. groups that have been configured as members of other groups) are **permanently logged in as incoming** and **permanently logged out as outgoing**.

Phones permanently logged in cannot log out.

- · Creating a group with an internal number.
 - Groups > Phone numbers > + New
- · Assigning an external number in the call distribution to the group.
 - Exchange lines > Exchange line routing > Exchange line routing incoming
- · Assigning members and properties to the group.
 - Groups > Telephone number > /
 - Groups > Profiles

Incoming Calls

A user is set up in the PBX on a person-specific basis with an internal call number. This allows a person to receive incoming calls on different, even changing, telephones.

Further help under <u>User</u>

Incoming calls for a group are controlled by membership in a group and logging in/out.

Further help under Groups

External incoming calls are controlled by the exchange line routing incoming.

Further help under Exchange line routing

Incoming calls from the door are controlled via door calls.

Further help under **Door functions**

Further control options, e.g. call forwarding or automatic call acceptance, are listed below.

With the call waiting function, incoming calls can be signalled during an ongoing call.

Further help under Call waiting

With Do-not-Disturb, incoming calls can be switched off if required.

Further help under **Do-not-Disturb**

Pick-up

Using pick-up you can take a call on your own phone that was originally sent to another internal telephone or to an own voice mailbox.

Further help under Pick-Up and Call Pick-Up

If the Pick up all calls to group members function is enabled, one member of the group can not only pick up calls to the group's phone number but also direct calls to individual group members.

Further help under Pick-up of all calls to group members

The Enquiry prior to pick-up function enables users to see which call they want to receive before they pick it up.

Further help under Preliminary pick-up query and monitoring

Call Forwarding

Use the Call forwarding function to forward calls independently of the current situation (immediately, on busy or on no reply). Different call forwarding options are available, depending on whether calls are to be forwarded for a single user, for a group or for external numbers.

Further help under Call forwarding

Use the Follow-me function to configure forwarding from the destination telephone.

Further help under Follow-me

The Multi-Path Call Forwarding function also signals incoming calls on another telephone, even an external telephone, although it does not forward the call.

Further help under Multi-path call forwarding

Call acceptance and announcements

The automatic reception function receives calls and connects callers with their required destination. If the Announcement before answering function is switched on, calls are placed directly in a waiting loop instead of being rejected if the line is busy. Alternatively, the destination device can be a voice mailbox, which either records a message or plays back an announcement, to provide information.

- Further help under <u>Announcement before answering</u>
- Further help under <u>Automatic reception</u>
- Further help under Voice mailbox

Call flow display

For incoming calls, a COMfortel D-series telephone displays which destination was called (external call number, group, forwarded destination).

- You will find more information about the indications in the display in the instruction of the telephone.
- Important: The call flow display is only available with feature level 2 or higher. If you want to use this function, please contact your dealer or system partner.
 - Further help under Feature level

InterCom

The InterCom function enables an audio connection to a system telephone from one or more telephones without someone having to actively receive a call (for example, in a medical practice).

Process

To set up InterCom on a phone number:

- 1. Dial the programming sequence for InterCom on an internal telephone.
 - The following rules/restrictions apply:
 - The destination can be either a single user or a group (several users).
- 2. The PBX system sends the SIP alert info for Auto Answer to the user's telephones.

The following rules/restrictions apply:

- Only users who have permission to use InterCom are taken into account.
- A SIP alert info is only sent to telephones that can evaluate it.
- 3. A free telephone (or more than one) automatically receives the InterCom call and enables the audio connection in one direction.
- Description of operation under <u>InterCom</u>

- Switching on the SIP alert info for Auto Answer for telephones that can, and are to, receive InterCom calls.
 - Telephones > Assignment > /

InterCom

- Switching on InterCom permission for the user whose telephones are to receive InterCom messages.
 - User > Phone numbers > /

Internal numbers

Individual devices and functions have to be assigned an internal number that is used to call them intenally. For this purpose, the following devices and functions receive internal numbers:

- User
- Groups
- Voice mailboxes
- · Fax boxes
- · Automatic receptions
- Conference rooms
- · Call on hold
- Door stations
- Announcement outputs
- Short-code dialling numbers

With a few exceptions, it must also be possible to call these devices and functions from outside. For this purpose, the internal numbers have to be assigned in the call distribution (exchange line routing incoming).

Further help under **Exchange line routing**

It is a good idea to create a dial plan for all devices and functions for the internal numbers before allocation and to transmit them to the PBX. Support is to be found in the following overview showing available and used phone numbers:

Monitoring and Overviews > Dial plan

The following rules have to be considered when creating internal numbers:

Rules for internal numbers

- Double allocation of a number is not possible.
- Two to four digit numbers in the range from 10 to 9999 are possible.
 - Door numbers: Only two to three digit numbers are possible. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers.
 - Call Parking: Only two to three digit numbers are possible. When put together with the ten possible park positions 0-9, this creates ten 3 to 4-digit phone numbers.
- Using numbers with few digits (two or three-digit numbers) inevitably restricts the possible supply of numbers requiring more digits. For example: the internal number 12 also occupies the internal numbers 120-129 and 1200-1299.
- The exchange line access number on the PBX is 0 in the factory settings. However, this can be changed. Changing the exchange line access number reduces the dial plan. For example: the exchange line access number 9 also occupies the internal numbers 90-99, 900-999 and 9000-9999.
 - Further help under **Exchange line request**
- Each account can be assigned its own exchange line access number for targeted dialling via a specific account. Also this exchange line access number reduces the dial plan.
 - Further help under Exchange line request
- Legal emergency numbers are entered without local area code. An emergency number from the range 10 to 9999 reduces the dial plan.

For example: the emergency number 112 also occupies the internal numbers 112 and 1120-1129.

Further help under Emergency calls

- · Creating a user with an internal number.
 - User > Phone numbers > + New
- · Creating a group with an internal number.
 - Groups > Phone numbers > + New
- Creating a voice or fax box with an internal number.
 - Voice mail and fax boxes > Phone numbers > + New
- Creating an automatic reception with an internal number.
 - Internal Devices > Automatic receptions > + New
- Creating a conference room with an internal number.
 - $^-$ Internal Devices > Conference rooms > + New
- Creating an internal base number for call parking.
 - Internal Devices > Call on hold
- Creating a door station or an announcement output with an internal number.
 - External Devices > Device wizard > + New
- Creating a short code number with an internal number.
 - Functions > Short-code number > + New

LAN TAPI

Overview of Functions
Installing TAPI driver (TSP)

Overview of Functions

For a CTI solution, the PBX can be connected to a local network (CTI server and CTI clients). In order to do so, a CTI software such as PBX Call Assist is required.

- You can purchase PBX Call Assist from the <u>Auerswald Shop/FONtevo</u> Shop.
- Further help under <u>Licences</u>

As an alternative, we recommend the following software manufacturers:

- ESTOS (estos)
- ilink (ilink)

The LAN-TAPI provides the TAPI 2.1 interface over the Ethernet port on the PBX.

A Microsoft server then distributes the functionality throughout the network (third party network connection) onto the separate Microsoft computers so that telephony software, e.g. PBX Call Assist or other TAPI applications can use the TAPI functions. The PBX is addressed by the server over the Ethernet interface. When a user wants to dial from the client PC, the command is forwarded to the server. The server signals this to the PBX which then establishes the connection.

LAN-TAPI supports the following phones connected to the PBX:

- · Analogue telephones
- Standard VoIP phones (SIP)

If a user is using TAPI on several telephones, they must specify which telephone is controlled by TAPI. This specification is not dynamic and does not change when the user switches from one telephone to another. Initially, the first telephone that was assigned to the user is the one that is

active. If the user wants to use a different telephone, they must actively set it as the TAPI telephone.

Functions

LAN-TAPI supports the following functions:

- Display of the called MSN (multi-company operation)
- Call forwarding (CFU, CFB, CFNR)
- Call deflection (forwarding before call acceptance)
- Transferring calls
- Splitting a call
- Monitoring (status display of the private branch exchanges)
- Pick-up
- Do-not-Disturb
- Blind transfer using ESTOS
- Important: The possible functions are dependent on the end devices/ applications used.
- Note: For controlling a telephone of the COMfortel D-series via LAN-TAPI, the CTI Access Token is required (from firmware version 1.4 of the telephone). The code given by the PBX for each phone is transmitted to the phone by provisioning.

- If required, enabling the use of PBX Call Assist.
 - Administration > Network > Server service > LAN TAPI

- If required, enabling the CTI Access Token for a telephone of the COMfortel D-series.
 - Telephones > Assignment
- Configuring the password for the network connection of the TAPI server and, if necessary, adjusting the network port of the TAPI server.
 - Administration > Network > Server service > LAN TAPI
- Installing TAPI driver (TSP).
 - Further help under <u>Installing TAPI driver (TSP)</u>

Alternative function on an internal phone

- · Set telephone as TAPI telephone
 - Description of operation under <u>LAN TAPI</u>

Installing TAPI driver (TSP)

The PBX has to be connected to a TCP/IP network using a server PC. The TAPI driver (TSP) belonging to the PBX has to be installed on this server PC.

- Important: If you are integrating the LAN TAPI into an existing network, contact the system administrator responsible for this. Making changes to an existing network may cause considerable malfunctions.
- You will find the current driver on the Internet under <u>Auerswald</u> <u>Support/FONtevo Support</u>.

Requirements:

- A current installation file (Setup.exe) saved on a data storage device (PC)
- Server PC, that meets one of the following requirements:
 - Windows 7 (32-/64-Bit)
 - Windows Server 2008 and Windows Server 2008 R2
 - Windows 8/8.1 (32-/64-Bit)
 - Windows 10 (32-/64-Bit)
 - Windows Server 2012 and Windows Server 2012 R2
 - Windows Server 2016
 - Windows Server 2019
- 1. Start the installation file (Setup.exe) on the server PC.
 - This starts the TAPI driver installation.
- 2. Proceed with the installation by clicking on **Continue**.

- 3. In the entry field IP address of the PBX enter the PBX's IP address.
- 4. In the entry field **Port** enter the network port of the TAPI server.
- 5. Enter into the entry fields **Password** and **Repeat password** the password for the network connection of the TAPI server
- 6. Click OK.
- 7. Click Close.
- 8. Restart the PC.

Licences

The operation of the system requires mandatory (Exception: Tryout mode) initialisation and the purchasing of the following licences:

- System activation (only COMtrexx Business and Advanced)
- · one or more user licenses

In addition, further licences can be purchased:

PBX Call Assist V1

User licences are offered in the form of floating user licences.

- For more information on floating user licenses, take a look at the Auerswald EULA. You will find this in the Internet under: Auerswald Support/FONtevo Support or here: Dashboard > Status information > Legal > End User License Agreement for Software (EULA)
- For initialisation during initial setup, take a look at the Instructions of your COMtrexx variation. You will find this in the Internet under:

 <u>Auerswald Support/FONtevo Support</u>.

The PBX regularly carries out a verification of the licences purchased and the legal activation. An overview of the licences purchased as well as links to the shop and tools for reinitialisation can be found here:**Administration** > **Licences**

- Important: The PBX requires access to the CIC server/server of the Voucher Center for verification. Make sure that the address and port are reachable.
 - Further help under Network
- For more information on data exchange with the CIC server and the server of the Voucher Center, take a look at the Privacy Policy of the COMtrexx. You will find this in the Internet under: Auerswald

<u>Support/FONtevo Support</u> or here: **Dashboard > Status information** > **Legal > Data protection**

Tryout Mode

The 14-day tryout mode enables a COMtrexx VM to be tested without the purchase of licences. In tryout mode, three user test licences are available for setting up users, which must be relicenced when exiting or expiring tryout. If more than three users are required, this requires the purchase of licences. Further use of the tryout mode ist then not possible.

Note: There is no tryout mode available for COMtrexx Business and COMtrexx Advanced. These PBXs always require the purchase of licences.

New installation of a PBX

When you newly install the PBX, your former licenses are still valid and do not have to be purchased again. The licenses are connected with the serial number of your PBX.

- 1. Reset your PBX.
- 2. Copy the initialisation code and log in into the license server.
- Under My devices all PBX that you have installed are listed. The PBXs are displayed with the serial number and the given name. Click on the serial number of the PBX you want to newly install.
- Click on Recreate certificate.
- 5. Paste the initialisation code and click on **Continue**.
- 6. Copy the answer of the license server into the empty field of the PBX registration.

Licences

Note: The recreated certificate contains among other information the serial number of your PBX. Using this serial number, your former licenses are available in the Voucher Center.

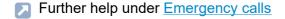
Locations

The entry of the location is mandatory for the first commissioning. With the country setting, the country code of the respective country is automatically adopted. Depending on this first location entry, also the emergency numbers of this country are adopted by the PBX.

If the location of individual accounts is different than the location of the PBX, these locations must be added and assigned to the accounts. By including a user in the call distribution of an account, a location is also assigned to the user. If several accounts have been assigned to a user in the call distribution, the location is assigned via the number presentation.

If the location of individual telephones is different than the registered locations, these locations must be added and assigned to the telephones. Softphones (also called soft clients) are very often not permanently bound to a location and are therefore created without a location by default.

The correctly assigned location is necessary, among other things, so that the PBX can properly initiate an emergency call.



When dialling a number without an area code, the phone's location (if set) or alternatively the user's location is dialled in advance. The PBX location is only used here if no other location can be identified.

- · Configuring further locations.
 - Exchange lines > Exchange line request/locations
- · Configuring locations differing from the standard for accounts.

- Exchange lines > Providers and Accounts > / > Settings > Network Provider
- If required, Configuring number presentation and thus the location separately for users.
 - User > Phone numbers > / > Exchange line settings
- If required, Configuring number presentation and thus the location configuration-dependent/time-controlled.
 - User > Profiles > / > Exchange line settings
- Configuring locations differing from the standard for phones.
 - Telephones > Assignment > /

Loudspeaker Audio Out

This function enables loudspeaker announcements over an active loudspeaker (for example, a loudspeaker system in a waiting room). In addition, a previously specified telephone number is called by an internal telephone. The call connection is immediately established after dialling the telephone number.

In order to make loudspeaker announcements, a device that can be used as announcement output (e.g. an a/b-Audiobox at a loudspeaker system) must be connected to the PBX. This announcement output must be created in the configuration of the PBX and assigned an internal number to call it. An existing relay can be used to switch the loudspeaker area of the loudspeaker system.



Description of operation under <u>Loudspeaker Audio Out</u>

- · Creating an announcement output with internal number.
 - External Devices > Device wizard
- Creating a relais and assigning it to an announcement output.
 - External Devices > Announcement outputs
- If required, making additional settings for the announcement output.
 - External Devices > Announcement outputs
- Switching on the authorisation for speaker announcements for the involved internal phones.
 - User > Phone numbers > / > Authorisations

Multi-path call forwarding

Multi-path call forwarding is used to signal internal and external calls, in addition to call distribution for an internal or an external number, e.g. on a mobile telephone. Multi-path call forwarding is also executed if the "main phone number" is, for example, busy.

Further help under Call distribution

If call forwarding for external numbers is additionally enabled, multi-path call forwarding might work only limited.

Further help under Call Forwarding for external numbers

For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply for the user/group are proved before a multi-path call is executed.

Further help under Exchange line routing

Use the Fixed Mobile Convergence (FMC) function if the mobile telephone is to be used as a fully functioning telephone by the user, instead of just having calls signalled on it.

Further help under FMC

- Configuring, enabling and disabling multi-path call forwarding separately for users.
 - User > Phone numbers > / > Reachability
- Configuring, enabling and disabling multi-path call forwarding separately for groups.

- Groups > Phone numbers > / > Reachability
- Configuring, enabling and disabling multi-path call forwarding configuration-dependent/time-controlled for users/groups.
 - User > Profiles > / > Reachability
 - Groups > Profiles > / > Reachability

Alternative functions on an internal phone:

- Configuring, enabling and disabling multi-path call forwarding for users.
- Configuring, enabling and disabling multi-path call forwarding for a group.
- Check activation.
 - Description of operation under <u>Multi-path call forwarding</u>

Music on Hold and Announcement

When transferring a call or in the case of a query call, a communication partner waiting in the background hears music on hold from the PBX, perhaps with an announcement played over the music.

Configuration

- Managing and assigning audio files for announcement and music on hold
 - Functions > Music on Hold/announcements > File management
 - Functions > Music on Hold/announcements > File assignment
- Switching on/off music on hold during the ringing period.
 - Functions > Music on Hold/announcements > File assignment
 - Internal Devices > Automatic receptions > /

Alternative functions on an internal phone

- Recording music on hold at music input
- · Deleting music on hold
- Record announcement
- Deleting announcement
- · Switching on/off music on hold and announcement and listening to it
 - Description of operation under <u>Music on Hold and Announcement</u>

MWI

MWI (Message Waiting Indication) is a SIP function which allows to signal new voice and fax messages on internal VoIP end devices. Signalling is possible for all voice mail and fax boxes for which the user in question is entered.

You can use this function with the following VoIP end devices:

Auerswald/FONtevo VoIP standard VoIP phones

The LED of a function key assigned to MWI is used for signalling (optional).

You will find more information in the instruction of your telephone.

Auerswald/FONtevo IP DECT phones

The signalling on the DECT handsets COMfortel M-5x0/M-7x0 is carried out via a symbol on the display (optional).

The setup is made using the web interface of the DECT base station WS-500x.

You will find more information in the instruction of the base station.

Standard VoIP end decives of other manufacturers

Signalling and configuration depend on the end device.

You will find more information in the instruction of the device.

Network

The PBX is connected to a local network and receives the IP address that is also used to access the integrated web server for configuration.

Please refer to the PBX Instruction for instructions about how to set up the network connection.

The PBX connects different communication nets with each other.

Further help under VolP

Only COMtrexx Business: By upgrading with a NET module, there are now two networks with different IP address ranges available.

Further help under Network interface (second)

Using the network, configuration data provided by the PBX is transerred to the connected IP telephones and DECT base stations.

Further help under Provisioning

In order to be able to securely connect remote devices to the PBX, the PBX offers an integrated OpenVPN server.

Further help under VPN

For a CTI solution, the PBX can be connected to a local network existing of CTI server and CTI clients.

Further help under LAN TAPI

Firewall releases for Auerswald services

If an existing firewall blocks the communication paths of the PBX to the Internet, releases must be granted for the following addresses and ports.

Network

Otherwise, it is not possible to use the associated services and thus the correct functioning of the PBX.

Name	IP addresses	Ports
cic.auerswald.de	81.14.169.81	443
vouchercenter.auerswald.de	81.14.169.81	443
Further help under <u>Licences</u>		
proxy.auerproxy.de	217.160.243.101	5060
	82.165.222.120	5061
	62.113.233.194	4433
	2001:8d8:1801:5f7::1	
	2001:8d8:1801:85f1::1	
	2a00:f48:2000:1009::2	
Further help under Softphone		
update.auerswald.de	81.14.169.81	80
		443
Further help under <u>Firmware Update</u>		

Protection against attacks through Fail2ban (IP block- and allowlist)

The PBX is protected against attacks from the network by Fail2ban. Fail2ban is software that monitors log files and searches them for predefined patterns.

If traffic from a particular IP address is judged as being too high - and therefore, probably malicious - this IP address is blocked. This means that access from this IP address to a service of the PBX is initially prevented for ten minutes (blocking time).

The blocked IP address is entered in the IP blocklist for the blocking time.

The PBX informs by means of a system message about:

- · System events detected by Fail2ban
- · Blocks triggered by Fail2ban
- Further help under <u>System messages</u>

You as the administrator must then set up suitable measures to protect the PBX within your infrastructure (e.g. via routers, bridges).

Remember that it depends on many factors whether an access is considered an attack. Even repeated »mistyping« when entering the password or a configuration error can lead to an entry in the IP blocklist.

If it is not a malicious attack, you can delete a blocked IP address from the IP blocklist again or transfer it to the IP allowlist. The IP allowlist makes it possible to exclude certain IP addresses from blocking. Enter here, for example, the IP address of the computer with which you normally configure the PBX.

IPv6 Mode

If IPv6 is configured, an IP address complying with version 6 of the Internet protocol can be assigned to the PBX. Due to the greater length of IPv6 addresses (eight blocks with four digits each, example: 3001:00FF:ABC0:0EAC:0001:0000:0000:000F), IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

Configuration

- · Configuring identification data.
 - Administration > Network > IP configuration > IP configuration
- If required, configuring IPv6 of the identification data.
 - Administration > Network > IP configuration > IP configurationIPv6 configuration > /
- · Configuring and testing DNS server.
 - Administration > Network > IP configuration > DNS configuration
- If required, configuring HTTP proxy.
 - Administration > Network > IP configuration > HTTP proxy configuration
- · Making network settings for the integrated web server.
 - Administration > Network > Server service > Web server configuration
- Managing block- and allowlist.
 - Administration > Network > IP block- and allowlist
- Viewing port overview.
 - Administration > Network > Ports

Alternative functions on an internal phone

- · Configuring and querying the IPv4 address of the PBX
- Configuring and querying the subnet mask of the PBX
- · Configuring and querying the gateway of the PBX

Network

- · Switching on/off the DHCP client of the PBX and querying the status
 - Description of operation under <u>Identification (IP Address, Subnet Mask, Gateway, DHCP Client)</u>

Network interface (second)

A second network interface can be configured in the PBX. Accordingly, two IADs (Integrated Access Devices) can be directly connected to the PBX and configured. Requirement is a COMtrexx Business with inserted NET module.

The following possibilities for using two network interfaces can be realised:

External VolP on LAN connection 2

- At the connections LAN 1 and LAN 2, an IAD is connected in each case.
- The external VoIP telephony is realised via LAN 2. There is no Internet connection at LAN 2.
- The internal VoIP telephony and Internet connection are realised via LAN 1.
- The data transmission can be optionally realised via a separate connection.

Internal VolP on LAN connection 2

- An IAD is connected to LAN 1.
- The data transmission and external VoIP telephony are realised via LAN 1.
- The internal VoIP telephony is realised via LAN 2.

VoIP completely on LAN connection 2

 At the connections LAN 1 and LAN 2, an IAD is connected in each case.

- The internal and external VoIP telephony are realised via LAN 2.
- The data transmission is realised via LAN 1.

- Configuring DHCP settings for the second network interface (NET module).
 - Administration > Network > IP configuration > NET module
- Configuring IAD.
 - Administration > Network > Gateways and routing > Internet Gateways
- · Selecting the used Internet access for the VoIP provider.
 - Exchange lines > Providers and Accounts > / > SIP
- Selecting network interface of the VoIP registrar.
 - Administration > VoIP > VoIP configuration > SIP

Number presentation

The number presentation outgoing defines which phone number is presented to the called person on an outgoing call. This is determined in the exchange line routing. Using the CLIP no screening function (availability on the account provided) allows a free assignment of the phone number to be transferred independently from the used connection.



The setting can be overridden by the individual setting **Transmit the own number** of the user/group.

The number presentation incoming defines which phone number is presented to the caller on an incoming call. This number may be different than the number actually dialled by the caller.

The internal number presentation on group calls defines which phone number is presented when a user makes an internal call as group member.

- If required, making settings about the format of the transferred phone number separately for the provider.
 - Exchange lines > Providers and Accounts > / > Number presentation (outgoing)
- Configuring number presentation incoming separately for accounts.
 - Exchange lines > Providers and Accounts > / > Phone numbers
- If required, activating CLIP no screening separately for accounts.

- Exchange lines > Providers and Accounts > / >Miscellaneous
- Configuring number presentation outgoing separately for users.
 - User > Phone numbers > / > Exchange line settings
- Configuring number presentation outgoing configuration-dependent/ time-controlled.
 - User > Profiles > / > Exchange line settings
- Configuring number presentation outgoing separately for groups.
 - Groups > Phone numbers > / > Exchange line settings
- Configuring number presentation outgoing configuration-dependent/ time-controlled
 - Groups > Profiles > / > Exchange line settings
- Configuring number presentation internal separately for groups.
 - Groups > Phone numbers > / > Settings
- Configuring number presentation internal configuration-dependent/ time-controlled.
 - Groups > Profiles > / > Settings

Phone number display (CLIP)

The phone number display (CLIP) enables the phone number of the caller to be shown on the display of a CLIP-capable analogue phone. The prerequisite is that the caller transmits a phone number.

A few CLIP-capable analogue devices support the recording of the date and time in conjunction with the caller list. The name can be displayed instead of or in addition to the phone number.

- Switching on CLIP information for the internal analogue devices.
 - Telephones > Assignment > /
- Switching on CLIP information for the analogue fax machines.
 - External Devices > Fax > /

Pick-Up and Call Pick-Up

Using pick-up you can take a call on your own phone that was originally sent to another internal telephone or to an own voice mailbox.

If the user is authorised to accept global calls, this applies to calls to all other users.

As an alternative, the authorisation for group calls and voice mailboxes can be configured. Here, the user is authorised to take calls from users that belong to his group. Furthermore, the user with this option is authorised to take over calls from those voice mailboxes of which he is the owner.

If a call has already been taken, the call can be taken using same procedure if the call take-over has been allowed by the user being called. This function is useful, for example, in the case of answering machines.



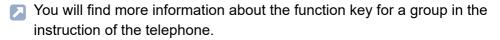
Description of operation under Pick-up

- · Switching on pick-up separately for users.
 - User > Phone numbers > / > Authorisations
- Switching on pick-up separately for users.
 - User > Phone numbers > / > Settings

Pick-up of all calls to group members

Using this function, a member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

Using for example a function key, not only calls to the phone number of the group, but also calls directly to the phone number of a group member will be signalled on the phone to the group member.



- Switching on pick-up of all calls to group members.
 - Groups > Phone numbers > / > Settings
- Configuring pick-up of all calls to group members configurationdependent/time-controlled.
 - ─ Groups > Profiles > / > Settings

Powering Down and Switching Off

If it is necessary to separate the PBX from the electric current (pull the power plug of the PBX out of the socket and also of the accessory equipment where required), the PBX will first have to be powered down properly.

Important: When shutting down the PBX, all of the connections to the web server, all calls and all data services are terminated. Unsaved data is lost.

Configuration

Proceed as follows for powering down and switching off the PBX:

- · Immediate powering down.
 - Administration > Maintenance > Update and Restart > Restart

Preferred exchange line

A preferred exchange line defines which one of the available accounts is preferred for initiating an outgoing external call. This is determined in the exchange line routing.



Further help under Exchange line routing

Preliminary pick-up query and monitoring

The PBX supports the SIP function BLF (in accordance with RFC 4235). BLF allows VoIP end devices to monitor other private branch exchanges and receive information about their calls.

Support of the SIP function BLF provides the following functions:

- Preliminary pick-up query: A preliminary pick-up query allows a
 user, even before a pick-up, to see which caller he would be picking
 up. Based upon the phone number of the caller, the user can then
 decide whether to pick up the call or not. As precondition, the called
 user or group must have enabled Preliminary pick-up query and
 monitoring.
- Monitoring or BLF (Busy Lamp Field): BLF allows VoIP end devices
 to monitor other internal users, groups, park positions of call parking,
 voice mail/fax boxes or door terminals and receive information
 about their calls or their busy status. In addition, the call partners of
 monitored users can be displayed. As precondition for the display
 of phone numbers, the monitored user or group must have enabled
 Preliminary pick-up query and monitoring.

You can use these functions with the following VoIP end devices:

- Auerswald/FONtevo VoIP phones (except COMfortel VoIP 2500 AB): Other private branch exchanges are monitored with the help of the LEDs of those function keys which are assigned to speed dialling.
 - You will find more information in the instruction of your telephone.
- VolP end decives of other manufacturers

Monitoring of other private branch exchanges on the same PBX depends on the end device.

You will find more information in the instruction of your telephone.

Description of operation under <u>Pick-up</u>

- Switching on preliminary pick-up query and monitoring for users and groups.
 - User > Phone numbers > / > Settings
 - Groups > Phone numbers > / > Settings
- · Configuring preliminary pick-up query and monitoring configurationdependent/time-controlled for users and groups.
 - User > Profiles > / > Settings
 - Groups > Profiles > / > Settings

Profiles

Profiles can be created for users, groups, and voice mail and fax boxes. These profiles control the behaviour by determining the properties. In this way, analogue behaviour can be reached via the configuration-dependent assignment of profiles.

When creating the profiles, it can be determined whether the individual settings from the profile can be changed via the properties of the user, group, or the boxes or not.

By using several configurations, the settings made in the profiles can also be changed time-controlled or manually.



Further help under Configuration switchover

- · Creating profiles for users and assigning them depending on the configuration.
 - User > Profiles > Profile management > + New > / Configure
 - User > Profiles > Profile Assignment
- Creating profiles for groups and assigning them depending on the configuration.
 - Groups > Profiles > Profile management > + New > Configure
 - Groups > Profiles > Profile Assignment
- Creating profiles for voice mail and fax boxes and assigning them depending on the configuration.

- Voice mail and fax boxes > Profiles > Profile management > +New > / Configure
- Voice mail and fax boxes > Profiles > Profile Assignment

Provisioning

Provisioning

Overview of Functions
Provisioning files
Creating provisioning file

Overview of Functions

Provisioning is the automated configuration of IP phones or DECT base stations via the provisioning server of the PBX from which the devices can transmit the necessary configuration data.

The device accepts the configuration data directly from a file distributed pver the network. A specific provisioning file is output for each device type supported in the PBX.

The **time of provisioning** is determined by two factors. For each device type, there is a defined weekday and point of time in the device-specific provisioning file in order to guarantee a regular provisioning. This is initiated by the PBX. Furthermore, a provisioning is initiated, if there are individual changes for the end device such as the new assignment of a user. In this example, the impuls is given by the requirements for the end device.

The **content of provisioning** is determined by the PBX. If there are changes such as a new assignment of a user to a telephone, the PBX send a message to the end device. It then requests the current data, the provisioning file for the device is updated.

First commissioning with activated auto provisioining

- 1. Auto provisioning is activated in the PBX.
- 2. The end device is connected to the PBX.
- 3. The PBX checks on demand of the device (multicast), if there is an appropriate template for this device type.
- 4. In the case of a match, the PBX transfers the URL of the provisioning server to the device.

5. The appropriate provisioning file is assigned to the device, the configuration data is transferred to the device.

First commissioning without auto provisioining

- 1. Auto provisioning is deactivated in the PBX.
- The end device is connected to the PBX.
- 3. The URL of the provisioning server is manually entered on the end device.
- 4. The PBX checks on demand of the device (multicast), if there is an appropriate template for this device type.
- 5. In the case of a match, the PBX transfers the URL of the provisioning server to the device.
- 6. The appropriate provisioning file is assigned to the device, the configuration data is transferred to the device.
- Note: There are further options for transferring the URL of the provisioning server to the end device such as using a DHCP server.

Requirements for provisioning

The following is necessary for provisioning:

- A provisioning server configured in the PBX
- In the PBX activated provisioning server
- If required, auto provisioning activated
- PBX supported telephones or DECT base stations (see table below) are configured as device type SIP telephone or DECT base
- Note: In case of used option Auto provisioning active: If there is another PBX with connected devices in the same network, it is

recommended to keep this option disabled. Auto-provisioning can only be switched on after all devices in the network have been checked for the necessity of automatic configuration (and the associated implementation of appropriate measures).

Note: If for the first commissioning of a telephone no user has been assigned, calling is impossible except for emergency calls.

Configurability	
COMfortel D-series (all models)	Yes
COMfortel 1400 IP	Yes, with restricted provisioning function
COMfortel 2600 IP	Yes, with restricted provisioning function
COMfortel 3600 IP	Yes, with restricted provisioning function
COMfortel WS-500x	Yes
Snom IP phones	Yes, with restricted provisioning function
Yealink IP phones	Yes, with restricted provisioning function

Characteristics

COMfortel D-series (all models)

These telephones automatically find the provisioning server in the delivery state.

With the templates contained in the assigned provisioning file, the functionality of the phone can be extended. These additional functions can be assigned to function keys.

Nou will find more information in the Advanced Information of the phone.

COMfortel 1400 IP / 2600 IP / 3600 IP

These telephones automatically find the provisioning server in the delivery state from firmware version 2.4.

COMfortel WS-500x

If a DECT base station is connected with the PBX and the device is switched on, the PBX checks on request of the device (multicast) the correct usage like it does with a telephone.

The DECT base station sends the request only three times after the connection is established (whereas telephones repeat sending signals again and again). For this reason and because of further specialties when configuring the base station and handsets, a special procedure is recommended.



Further help under DECT system

Excluding devices from provisioning

There are the following options to exclude certain devices from provisioning enabled in the PBX.

The first setup of a phone as SIP telephone:

- Using the device type SIP telephone
- Assigning no provisioning file
- Assigning a user
- · Then assigning the user permanently (thereby setting him as the owner)

The first setup of a phone as SIP soft client:

Setting up the device as type SIP soft client

If the setup of the telephone has already been made:

- · Deleting the provisioning file assigned to the device
- Permanently assigning a user that has already been assigned, if necessary (thereby setting him as the owner)
- · Switching off the function Roaming User for this user, if enabled

Initial Commissioning of a DECT base:

- Using the device type DECT base
- Assigning no provisioning file
 - Further help under DECT system

- Creating new telephone or DECT base station with provisioning specifications
 - Telephones > Assignment > New
- Assigning MAC address, copying address of the provisioning server.
 - Telephones > Assignment > /
- Editing, creating, importing or exporting provisioning file.
 - Telephones > Provisioning
- Configuring options for the provisioning.
 - Administration > Network > Server service > Provisioning
- · Making provisioning settings on the phone.
 - Configuring type of provisioning

- If required, entering the address of the provisioning server, starting with https://.
 - You will find more information in the instruction of your telephone.

Provisioning files

Provisioning files contain configurations for phones or DECT base stations. For Auerswald/FONtevo IP phones and DECT base stations. these provisioning files are *.xml or *.cfg files, built after a defined scheme. They contain e.g. account, provider or function key information.

Usually, a provisioning file is assigned to all devices of a type. If an IP phone or DECT base station requests a provisioning, the PBX replaces the data of the provisioning file with suitable data content.

A provisioning file that was not generated in the PBX (e.g. via a backup of the telephone), is typically applicable for only one user, as the file contains personal data such as account information. Provisioning data is userspecific if they contain at least user name and password of the user at the VoIP account.

Provisioning templates

All provisioning files provided in the PBX are device templates with a defined scheme. These files cannot be changed, deleted or exported.

Another provisioning file can be made from these templates in two different ways so that for example certain devices of a device type can be configured separately.



Further help under Creating provisioning file

Variables as placeholder

Variables for the parameters to be provisioned are created as placeholders in the configuration files provided by the PBX.

Provisioning files

Provisioning examines a file to be delivered to the phone. The recognised variable names are replaced by the values from the PBX configuration. The changed file is then sent to the device to be provisioned.

Creating provisioning file

In order to individually adjust the configuration of certain phones or DECT base stations, individual provisioning files can be created and assigned to the devices.

Note: You will find the account IDs used in the PBX and for the exchange lines in the file editor under Telephones > Provisioning > Editor > / > Account IDs.

From a device template (template)

In the web interface under **Telephones > Provisioning**:

- 1. Select + New.
- 2. Assign a file name.
- 3. Select a device type.
- 4. Adjust the file content.
- 5. Assign the provisioning file to certain devices as requested.

From an already used and maybe modified file

In the web interface under **Telephones** > **Provisioning**:

- 1. Select **Editor** for a data entry for the according device type.
- 2. Select | Save as.
- 3. Assign a file name.
- 4. Adjust the file content.
- 5. Assign the provisioning file to certain devices as requested.

From a backup of the telephone

Using the web interface of a telephone of the D-series or a COMfortel 1400 IP/2600 IP/3600 IP, a configuration can be created and afterwards, the configuration file that is supposed to be distributed to the phones selected.

- 1. Open the web interface of the telephone.
- 2. Go to the part of export of phone configuration and export the *.xml file to your PC.
 - You will find more information in the Advanced Information of the phone.
- 3. Open the *.xml file in a text editor such as Notepad++.
- 4. Select the layers you want to distribute to other phones by deleting the unnecessary layers.
- 5. Adjust the entries in the file as requested.
- 6. Import this file into the PBX.
- 7. Assign the provisioning file to certain devices as requested.

Query and Transfer

The PBX requires specifications for the transfer and the query call to be executed as desired.

The correct exchange line routing rule set for the user must support the functions accordingly.

Further help under Exchange line routing

Three types of transfer are available:

- · Transfer the call with an announcement
- Transfer the call without announcement (blind transfer)
- Query and 3rd Call

Process

Query and Transfer are carried out as follows:

Query call

- 1. Dialling the call number to which the enquiry call is to be made from an existing call.
- 2. The first call partner is put on hold and hears the on-hold music.
- 3. The call is accepted and the connection to the called partner is established.
- 4. You have the following options:
 - You are switching between the two call partners (splitting).
 - You end the connection with one of the two call partners and are then connected with the other call partner in each case.

- You hang up and your two conversation partners are then connected.
- One of your two call partners hangs up and you are still connected to the other call partner.

Transfer the call with an announcement

- 1. Dialling the call number to which the call should be transferred to from an existing call.
- The first call partner is put on hold and hears the on-hold music.
- 3. The call is accepted and the connection to the called partner is established.
- 4. You hang up.
- The connection with the previous call partner is established.

Transfer the call without announcement (blind transfer)

- 1. Dialling the call number to which the call should be transferred to from an existing call.
- 2. The first call partner is put on hold and hears the on-hold music.
- 3. You hang up.

The following rules/restrictions apply:

- After hanging up, the call number of the person to be transferred is transmitted to the transfer destination
- COMfortel SoftPhone is transfer destination: Your call number is still displayed at the transfer destination. The call number of the person being transferred is only visible after the call has been accepted.
- 4. You have the following options:

- The call is accepted and the connection between the called partner and the first call partner is established.
- The call is not answered within 60 seconds and is returned to you. Your phone rings. If you do not accept the call within 60 seconds the call will be disconnected. If your own telephone has become busy in the meantime, the procedure will already be canceled after the first 60 seconds and the call will be disconnected.

Query and 3rd Call

- 1. Dialling the call number to which the enquiry call is to be made from an existing call.
- 2. The first call partner is put on hold and hears the on-hold music.
- 3. The call is accepted and consultation is held.
- Dialling another number.
- 5. The query call is put on hold.
- The call is accepted.
- 7. You have the following options:
 - You end the third call and the connection to the last active call is established.
 - You end the third call and the connection to the last held call is established.
 - The second and third call partners are connected to each other and the connection with the first call partner is established.
 - The third conversation continues and the first and second calls are disconnected.
- Description of operation under <u>Query and Transfer</u>



Description of operation under <u>Query and 3rd Call</u>

- Configuring hook flash time for an analogue phone.
 - Telephones > Assignment > /

Relays (actuators)

Relays (actuators)

Overview of Functions Relay Operating Mode

Overview of Functions

The PBX supports the connection and control of various types of relays. A relay can be part of a connected door terminal or another connected device (e.g. an IP switch relay). Relays can have various functions on the PBX.

- Further help under Relay Operating Mode
- Note: The internal system relay with the identification number 900 (factory setting) has a special position. It can only be used in the operating type automatic configuration switchover to switch it on or off via phone. For this purpose, the relay is being switched on or off.
 - Description of operation under Relays

- · Creating an IP switch relay.
 - External Devices > Device wizard
- Creating a relay.
 - External Devices > Relays (actuators)
- Configuring the operating mode for the relays.
 - External Devices > Relays (actuators)
- · Configuring relays.
 - External Devices > Relays (actuators) > /
- Switching on authorisation separately for switching of relays for the involved internal phones.
 - User > Phone numbers > / > Authorisations

- Switching on authorisation configuration-dependent/time-controlled for switching of relays for the involved internal phones.
 - User > Profiles > / > Authorisations

Alternative functions on an internal or external Phone

- · Switch relay
 - Description of operation under <u>Relays</u>

Relay Operating Mode

Relays can have various functions on the PBX.

You can configure the following operation modes:

Speaking

The relay is assigned to a door terminal or an announcement output.

The PBX switches on the relay as soon as an audio connection has been established between the PBX and the assigned device (e.g. an a/b-Audiobox).

The PBX switches the relay off as soon as the audio connection is terminated.

Possible use:

- Switching an LED of the door terminal
- Switching the loudspeaker area of the loudspeaker system for the announcement output

Door opener

The relay is assigned to a door terminal.

The PBX switches on the relay as soon as an internal user triggers door opening for the assigned door terminal.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

· Switching a door opener

Light

The relay is assigned to a door terminal.

The PBX switches on the relay as soon as an internal user triggers light switching for the assigned door terminal.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

Switching a staircase light

PC relay

The relay is assigned to a user.

The PBX switches on the relay as soon as the assigned user is being called.

Alternatively, switching can be triggered manually by phone.

The PBX switches off the relay after the defined switching duration.

Possible use:

· Switching a PC

Second ringer

The relay is assigned to a user, a group or a door call.

The PBX switches on the relay as soon as the assigned user or group is being called, or at the start of a door call.

The PBX switches off the relay at the end of the call.

Possible use:

· Switching a second ringer

Configuration-dependent

The relay is assigned to one or more configuration/s (day, night, etc.).

The PBX switches on the relay as soon as an assigned configuration is switched on.

The PBX switches off the relay as soon as a configuration to which it is not assigned is switched on.

Alternatively, switching on/off can be triggered manually by phone.

Auxiliary relay (secondary)

The auxiliary relay is assigned to a primary relay.

The PBX switches the auxiliary relay together with the assigned relay.

Manual relay

The PBX does not automatically switch on and off the relay.

Switching on/off can only be triggered manually by phone.

Reserved call channels

If a group of users is to be able to have an external call at any time, call channels can be reserved for this group. The reservation can only be made on one account and the associated provider. These call channels are then only available to users who are logged in this group as outgoing.

The following rules/restrictions apply:

- · Dynamically used channels are not offered.
- The maximum number of channels available for the account relates to the number that has been set on the assigned provider.
- If a call channel is to be reserved for a single user, the user must be assigned to a group alone.

- Reserving VoIP channels for external calls.
 - Administration > VoIP > VoIP configuration > Distribution of the VoIP channels
- Configuring maximum number of provided VoIP channels separately for the provider.
 - Exchange lines > Providers and Accounts > Name of the provider | / > RTP
- · Configuring reserved call channels separately for a group.
 - Groups > Phone numbers > / > Reachability
- Configuring reserved call channels configuration-dependent/timecontrolled.
 - Groups > Profiles > / > Reachability

Restart

A restart (reboot) is when you cause the PBX to reboot while it is already functioning. The configuration of the PBX is not deleted.

A restart is required, for example, in the following situations:

- After the operating software on the PBX has been updated (firmware update)
- · After the operating software on the PBX crashes
- After a change to the configuration that cannot be applied when the PBX is in operation

There are also various options for performing a restart yourself, if necessary.

Automatic restart

An automatic restart is executed after the following functions have been carried out:

- · Instant firmware update
- Restoring firmware (reboot)
- Restoring Factory Settings
- Restoring Configuration Data from a Restoration Point
- · Resetting configuration
 - Note: Resetting the configuration requires only a limited restart. Only a part of the PBX functions is restarted. All calls are terminated and all data services are closed. The web server remains unaffected by this.

Manual restart

Proceed as follows for a manual restart:

- Note: During an immediate restart, no consideration is made of existing calls which means that all connections to the web server, all calls and all data services are closed immediately. Unsaved data is lost.
 - Immediate restart.
 - Administration > Maintenance > Update and Restart > Restart

Alternative function by pressing a key on the PBX (only COMtrexx Business)

- · Initiate reboot
 - Description of operation under <u>Button</u>

Roaming User

The Roaming User function enables users who, for example, work in a large office with workplaces that are not assigned to specific people (i.e. "hot desking") to access any telephone that is configured in the PBX.

The user has the known access with his personal settings and authorisations.

If the Roaming User function is used in combination with the central caller list, the user is given access to all their calls (incoming and outgoing). This requires a COMfortel D-series telephone.

- Further help under User
- Further help under Central caller list

Accessing telephones

A telephone that has not been assigned to a particular user can be used by any user who has roaming user authorisation. Users with this authorisation can log onto this telephone at any time.

Even if the telephone is assigned to a specific user, they can log out of it to free it up for another user. In this case, the second user must also have roaming user authorisation.

If a telephone is permanently assigned to a user, that user is its owner. If the owner temporarily logs out of this telephone, other roaming users will have no access to it. The owner does not need roaming user authorisation

Use case open-plan office

The office has several freely assignable workstations with telephones. These telephones have not been assigned to a user.

A user who has roaming user authorisation can log onto one of these telephones. When they change workstation, they simply log out of the telephone to free it for other users. On another day, the user can log onto a different telephone.

Use case office / home office

A user's telephone at their workplace office and at their home office are both configured in the PBX.

The user must log on to one of these telephones to ensure that this is the only telephone that rings. If the user is also configured as the telephone's owner, they do not need roaming user authorisation.

Process

This is the procedure for logging into and out of the telephone:

1. The user logs into the phone of their choice by entering their PIN and user phone number.

The following rules/restrictions apply:

- The user must either be configured as the telephone's user or have authorisation as a roaming user.
- 2. To change to a different telephone, the user then simply logs on to that phone directly.

The following rules/restrictions apply:

Roaming User

- When the user starts using a different telephone, they are automatically logged out of the telephone they were using previously. If they do not want to do this, the user can simply log onto an additional telephone.
- 3. To free up one or more telephones for other users who have roaming user authorisation, the user can actively log out of one or more telephones.

The following rules/restrictions apply:

- This procedure does not log the user out of a mobile telephone that is set up to use FMC.
- When they log out, the local caller list on a COMfortel D-series telephone is deleted.
- Description of operation under Roaming User

- Switching on roaming user authorisation separately for participating users.
 - User > Phone numbers > Authorisations
- Either assign the participating telephones to a user or release the telephones.
 - Telephones > Assignment

Service data

In case of a malfunction of the PBX, a log can be recorded. This can then be forwarded to the specialised departments (Serice, Development) for troubleshooting. The following protocol options are available:

- Recording a network trace in PCAP format to control the network data stream
- Logging of the control and management information of the PBX connections
 - in PCAP format e.g. via the PC programme Wireshark
 - via a syslog server

In addition, Ping can be used to check if the PBX has an existing network connection.

Data protection for network diagnosis trace

Important: Note the following legal notes prior to recording a network data stream.

The recorded network data stream may contain the following components:

- Data streams of other computers in the network
- Connection data sets with destination, source, and time information of a telecommunication connection
- The spoken word of both ends of a telecommunication connection
- Configuration data of the PBX and connected phones, as well as the corresponding PINs and passwords

Before the backup is being saved, all users to the PBX must be informed that these contents will be forwarded and about who will

use the data received. It is essential to store the data securely and in accordance with the legal requirements. If you want to make the backup available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled

Network diagnosis trace process

Requirements:

- Pop-up blocker which is disabled in the browser or a defined exception rule
- Start the trace under Administration > Service > Network Diagnosis
 > > Start Trace.
- 2. If the malfunction can easily be reproduced, try to trigger it.
- 3. Finish the trace by clicking **Stop Trace**.
- 4. Save the file.
- Please refer for evaluation to <u>Auerswald Support/FONtevo Support</u>.

Data protection for logging

Important: Please note the following legal information before approval: After activation, any computer in the LAN can read and record the log records of the PBX from the time of activation without any further password protection.

The log records can contain the following components:

 Connection data sets with destination, source, and time information of a telecommunication connection

Service data

- Configuration data of the PBX and connected phones, as well as the corresponding PINs and passwords
- · Address books of the PBX

Before any recording, all users in the telecommunication connection must be informed that these contents will be recorded and must also be informed who will use the recorded data. If you want to make the recorded data available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Process for logging in PCAP format (here with Wireshark)

Requirements:

- Logging activated for logging systems that support PCAP format
 - Administration > Service > Logging > Logging activated/Logging system
- Installed PC programme Wireshark
- 1. Start the command line.
- 2. Enter the following:
 - start wireshark -k -i TCP@IP address:Port
 - The PBX's IP address: IP address
 - Port for logger output in Wireshark:Port
 - Example: start wireshark -k -i TCP@192.168.0.240:42231
- 3. Click Enter.
- 4. Wireshark starts the logging.

Service data

- Note: The log output by Wireshark is coded.
- Please refer for evaluation to <u>Auerswald Support/FONtevo Support</u>.

Short-code number

With the aid of short-code dialling numbers, longer digit sequences can be dialled more easily. Alternatively, sequences of macro commands can be executed.

Both options can be used, for example, to simplify frequently used digit sequences for operating or programming the PBX.



Not supported::

- Replacing an internal call number with another internal call number.
- Integration of a macro in another macro.

Process

Dialling a short-code number is carried out as follows:

Short-code number with simple digit sequence e.g. for logging out of all groups

Phone number 123, Macro sequence ##8*400#

- 1. Dial the short-code number 123 on an internal telephone.
- 2. The PBX dials ##8*400#.

Short code number with sequence of macro commands, e.g. for calling a service with automatic reception

Phone number123, Macro sequence Z080012345CW2Z2

1. Dial the short-code number 123 on an internal telephone.

2. Command **Z080012345**: The PBX dials/starts a call to the telephone number 080012345.

The following rules/restrictions apply:

- While the PBX is processing the macro commands, there is only a one-way audio connection. It is possible to listen in on the phone, but not to actively participate.
- Hanging up prematurely on the telephone cancels the sequence of macro commands.
- If there is a problem/error when evaluating the macro commands, the PBX terminates. You will hear a busy tone for 2 seconds.
- 3. Command **C**: The PBX is waiting for the connection to be established.
- 4. Command **W2**: The PBX waits an additional 2 seconds.
- 5. Command **Z2**: The PBX transmits the digit 2 via DTMF.
- 6. After processing the macro commands, the connection is completely established.

The following rules/restrictions apply:

- The connection can be continued and can be terminated by hanging up on the telephone.
- If the sequence of macro commands ends with H for hang up, the PBX automatically hangs up after processing the macro commands. A complete connection is not established.
- If subsequently dialled digits are not to be saved as a macro command, they can be dialled directly after dialling the short code number. The PBX first processes the sequence of macro commands and then dials subsequently selected digits.

- Configuring short-code dialling numbers.
 - Functions > Short-code number

Softphone

This PBX supports the operation of the Auerswald COMfortel SoftPhone and softphones from other manufacturers (also called soft clients).

COMfortel SoftPhone

Important: The COMfortel SoftPhone is available licence-free for a limited period of time. For permanent use, please contact your dealer or system partner.

The COMfortel SoftPhone is a multi-platform client for Windows, Android and iOS.

The COMfortel SoftPhone is provisioned via the PBX.

The registration information for the COMfortel SoftPhone can be sent to the user by e-mail.

Thanks to the proxy service, the COMfortel SoftPhone is also available on the smartphone regardless of the location. No VPN connection is needed.

- Further information on the COMfortel SoftPhone operation can be found in the COMfortel SoftPhone Operation Manual. You will find this in the Internet under: Auerswald Support/FONtevo Support.
- Important: For the provisioning and proxy service, the PBX requires access to the Auerproxy server. Make sure that the address and port are reachable.
 - Further help under Network
- For more information on the proxy service via the Auerproxy server, please refer to the COMtrexx data protection declaration. You will find this in the Internet under: Auerswald Support/FONtevo Support or here: Dashboard > Status information > Legal > Data protection

Requirements for the transfer of e-mails

- · Configured outgoing mail server
 - Further help under E-mail
- · Configured DNS server and, if necessary, an HTTP proxy server
 - Further help under Network

- Activate the proxy service to use the COMfortel SoftPhone.
 - Administration > VoIP > VoIP configuration > Proxy server for COMfortel SoftPhone > Use proxy
- Set up the phone and assign it to a user. COMfortel SoftPhone: Enter an e-mail address for the transmission of the registration information.
 - Telephones > Assignment > + New
 - Type: SIP soft client
 - User
 - Send registration link
 - Note: Softphones are very often not tied to a location and are therefore created by default without a location. The correctly assigned location is necessary, among other things, so that the PBX can properly initiate an emergency call.
 - Further help under **Emergency calls**
- · Install Softphone in the device.

Softphone

- The installation file for the COMfortel SoftPhone can be found on the Internet at <u>Auerswald Support/FONtevo Support</u>. COMfortel SoftPhone Mobile: The clients are installed via the Google Play Store or Mac App Store.
- Enter user name and password in the softphone.
 - Note: The registration information for the COMfortel SoftPhone is emailed to the user when the phone is set up. Sending can be restarted later (e.g. after changing the user):
 - Telephones > Assignment > / > Send registration data again
 - Note: The registration information can be copied here:
 - Telephones > Assignment > /
 - SIP user name

Switch input

If a device that is suitable as switch input (e.g. a door terminal with door bell buttons) is connected to the PBX, it can then be used to trigger a door call.

- · Creating and configuring a switch input for a door call.
 - External Devices > Door stations

System messages

System messages are used to inform administrators and users about system events such as operating errors.

The following different types of system events are possible:

- system information, e.g. automatic memory clean-up and memory check (Voice Mail/Fax function), switch between configurations, switch relays on/off, send system e-mails, register on the web interface
- simple malfunctions, e.g. automatic memory clean-up and memory check (Voice Mail/Fax function) errors or errors when sending softphone registration data
- serious malfunctions, e.g. fan fault (only COMtrexx Business), alarm temperature exceeded (only COMtrexx Business)
- security warnings, e.g. failed login attempts, entries in the IP blocklist, Admin password change

Occuring events are logged and can be looked at under:

Administration > Maintenance > System messages

E-mails with information about the events can also be sent.

Requirements for the transfer of e-mails

- Configured outgoing mail server
 - Further help under E-mail
- · Configured DNS server and, if necessary, an HTTP proxy server
 - Further help under Network

- · Set up the sending of system e-mails and event logging.
 - Administration > Maintenance > Message Configuration
 - Voice mail and fax boxes > Basic Settings > Basic settings for all voice mail and fax boxes
- Configuring E-Mail Transfer separately for voice and fax boxes.
 - Voice mail and fax boxes > Phone numbers > / > Box settings

Templates

Templates are provided by Auerswald in the form of XML files that can be used in the PBX. They provide a basic configuration and can be imported, exported and partly edited using the web interface.

Provider template

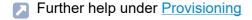
A provider template sets a configuration for the PBX that makes the configuration with a certain provider easier and allows a preferably problem-free operation. It describes how the PBX works with the provider.

As far as required from the provider, the template can be edited using the web interface.



Provisioning templates

A provisioning template is adjusted to the manufacturer of the end device and contains for example information about the account, the provider or function keys.



Device template

Different device templates for different devices and manufacturers are provided. In addition, user-defined templates can be selected.

A device template contains device-specific information such as how a device is controlled or how many outputs a device has.

Further help under <u>Loudspeaker Audio Out</u>

Templates

- Further help under Relays (actuators)
- Further help under **Door functions**

Time control

Time-controlled configurations are toggled according to the PBX internal clock. A configuration applies until another configuration (manual or automatic) is enabled, also beyond the daily change.



Further help under Configuration switchover

Tones

The PBX supports different tones on the receiver in order to signal the various operating states on the PBX.

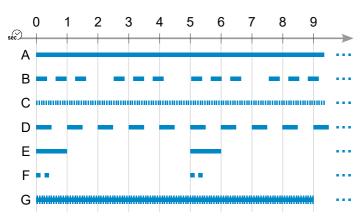
Signaling scenarios

- · External dial tone (long dial tone):
 - After dialling the exchange line access number, when a free external line is available – dialling a telephone number is possible
- Internal dial tone (private branch exchange dial tone):
 - After picking up the receiver or after pressing the Flash or Query key, when a free line is available – dialling a telephone number is possible
 - Note: On standard VoIP phones, a dial tone is generated by the phone itself, even when no free line is available.
- Confirmation tone:
 - At the completion of programming, when the programming has been successfully executed
- · Busy tone:
 - After picking up the receiver, when no free line is available or an authorisation is missing
 - After dialling a phone number, when the desired connection is busy or the network is overloaded or an authorisation is missing
 - At end of call
 - During programming, when the programming was unsuccessful or an authorisation is missing

- During a Call Through call, when the telephone number transferred from the mobile phone is not permitted to use Call Through
- Call tone:
 - During the call phase
- Call waiting tone:
 - During a call (when call waiting has been enabled), when another call comes in – it is possible to accept or reject the additional call
- · Changing tone:
 - During a Call Through call, when the telephone number transferred from the mobile phone is permitted to use Call Through - dialling a telephone number is possible
 - During a remote programming call, when the connection for remote programming is available – programming is possible

Display of signaling types

The tones are signalled as follows:



A Permanent dial tone (425 Hz)

Tones

- B Private branch exchange dial tone (425 Hz)
- C Confirmation tone (425 Hz)
- D Busy tone (425 Hz)
- E Call tone (425 Hz)
- F Call waiting tone (425 Hz)
- G Changing tone (697/1633 Hz)

User

A user is set up in the PBX on a person-specific basis. The user is assigned a phone number, PIN, password and certain properties.

One user can be temporarily or permanently assigned to up to five telephones. This then means that they can, for example, use a telephone that is configured at their office workplace, a telephone that is in their home office and a DECT telephone in the company building. All the telephones have the user's personal settings and permissions and can be accessed using their phone number. This enables the PBX to identify whether a user's telephone is occupied and inform the caller about this.

The FMC function can also be used to set up a mobile telephone as the user's telephone.

Further help under FMC

When a user is permanently assigned to a telephone, they become the owner of that telephone. If the user has this status, they can then log onto and out of the telephone on a temporary basis, for example, to prevent the telephone in their home office from ringing when they are at their office workplace.

Description of operation under <u>Roaming User</u>

If the user is authorised to use the roaming user function, other special features apply to access to telephones.

Further help under Roaming User

If used in combination with the central caller list, the user is given access to all their calls (incoming and outgoing). This requires a COMfortel D-series telephone.

Further help under Central caller list

If a user is using TAPI on several telephones, they must specify which telephone is controlled by TAPI. This specification is not dynamic and does not change when the user switches from one telephone to another. Initially, the first telephone that was assigned to the user is the one that is active. If the user wants to use a different telephone, they must actively set it as the TAPI telephone.

Further help under LAN TAPI

Several users can be grouped, in order to be available by the same phone number and/or to acquire additional properties.

Further help under Groups

Configuration

- · Creating a user with an internal number.
 - User > Phone numbers > + New
- Assigning an external number in the call distribution to the user.
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Call distribution for
- · Assigning properties to the user.
 - User > Phone numbers > /
 - User > Profiles
- · Making general settings for all users.
 - User > Functional Overview > Global settings
- · Assigning users to the telephones.
 - Telephones > Assignment

Alternative function on an internal phone

User

- Set telephone as TAPI telephone
- Description of operation under <u>LAN TAPI</u>

Voice mail and fax boxes

Overview of Functions Voice mailbox Fax box Fax Transfer

Overview of Functions

The PBX provides voice mail and fax boxes. The number of available boxes and the voice mail and fax channels for them depends on the product variant of the COMtrexx.

You will find further information in the instructions of your device.

The messages are managed separately for voice mail an fax boxes under **User data > Messages**.

Voice mail and fax messages can be forwarded as e-mail.

Further help under E-mail

Memory

The following is stored on the internal memory:

- · Incoming voice messages
- Incoming faxes
- Announcements for the voice mailboxes
- Language files

- · Making basic settings for the storage management.
 - Voice mail and fax boxes > Basic Settings > Basic settings for all voice mail and fax boxes

Voice mailbox

As an answering machine the voice mailbox accepts incoming calls and stores recorded messages as audio files (*.wav). Up to 500 messages per box are possible.

Announcement texts can be assigned for each box.

On certain phones, new voice messages are signalled by the LED on the message key (optional).

You will find more information in the instruction of your telephone.

In addition, the user of a voice mailbox can be informed about new voice messages by e-mail or by SIP MWI (optional).

Further help under MWI

The voice messages of a voice mailbox can be queried by means of 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.

Process

Requirements:

- Knowledge of the PIN for accessing the corresponding voice mailbox remotely
- · DTMF dialling support on the external telephone
- 1. Call the voice mailbox.
 - Note: 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 greater than 50 seconds).

- 2. Select * and then the user or remote access PIN and #.
 - Note: If the internal telephone used for the query is the owner of the voice mailbox, it is not necessary to enter a user PIN or remote access PIN.

Different announcements guide through the remote access menu.

Description of operation under <u>Voice mailbox</u>

- · Configuring a voice mailbox.
 - Voice mail and fax boxes > Phone numbers
- Switching on/off voice mailbox for a user/group.
 - User > Phone numbers > / > Voice mail and fax boxes
 - Group > Phone numbers > / > Voice mail and fax boxes
- Configuring announcement language, remote access, message forwarding E-mail transfer among other things.
 - Voice mail and fax boxes > Phone numbers > / > Box settings
- Configuring storage management separately for each voice mailbox.
 - Voice mail and fax boxes > Phone numbers > / > Box settings
- · Configuring announcements.
 - Voice mail and fax boxes > Phone numbers > / >Announcements

- Voice mail and fax boxes > Phone numbers > / > Call acceptance/announcement
- · Configuring readiness and call acceptance.
 - Voice mail and fax boxes > Phone numbers > / > Call acceptance/announcement

Alternative functions on an internal phone:

- · Switching on/off readiness
- Switching on time control for readiness
- · Recording, listening to, deleting announcements for voice mailboxes
 - Description of operation under <u>Voice mailbox</u>

Fax box

A fax box accepts incoming faxes and saves them as *.pdf files. Up to 500 messages per box are possible. The maximum length of a fax is 50 pages.

The fax receiving function is supported with maximally 9.6 kbps (V.29).

- · Configuring a fax box.
 - Voice mail and fax boxes > Phone numbers
- Switching on/off fax box for a user/group.
 - User > Phone numbers > / > Voice mail and fax boxes
 - Group > Phone numbers > / > Voice mail and fax boxes
- Configuring fax ID, fax transmission and E-mail transfer among other things.
 - Voice mail and fax boxes > Phone numbers > / > Box settings
- · Configuring storage management separately for each fax box.
 - Voice mail and fax boxes > Phone numbers > / > Box settings
- Configuring readiness and call acceptance.
 - Voice mail and fax boxes > Phone numbers > / > Call acceptance/announcement

Fax Transfer

Overview of Functions
Installing fax printer driver on a PC
Configuring fax printer driver
Send Fax to a Recipient
Send Fax to several Recipients

Overview of Functions

The fax function of the PBX supports fax transfer in combination with a corresponding PC application. The required **COMpact/COMmander Fax** printer driver has to be installed on the PC used.

The 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 destination of the fax is not available (busy or no reply), two further dial-up attempts will automatically be made by the PBX.

Installing fax printer driver on a PC

- You will find the fax printer driver **COMpact/COMmander Fax** on the Internet under <u>Auerswald Support/FONtevo Support</u>.
- 1. Download the printer driver and start the installation.
- 2. Select the desired language and click on **OK** and **Continue**.
- 3. Read and accept the license agreement. Select Continue.
- 4. Select Continue installation and Finish.

Configuring fax printer driver

Before sending the first fax using the installed printer driver **COMpact/ COMmander Fax**, at least one fax box of the PBX has to be configured in the printer driver.

- In order to open the printer driver, print a test document (e.g. word document) via the printing function of the PC program. In the printer selection dialog, select the printer driver COMpact/COMmander Fax.
- If necessary, enter in the entry field Fax box name a name for identification.
 - Note: It makes sense to use the same name that has been given to the fax box in the web interface of the PBX.
- 3. In the entry field **User/group number** enter the internal number of the user of the correponding fax box.
 - Note: Entering a group number is not possible.
- 4. In the entry field **User Password** enter the password of the corresponding user.
 - Important: By activating the checkbox Display PIN, the entered password is displayed unencrypted. As for security reasons, deactivate this function immediately after editing.
- 5. In the entry field **Network address** enter the PBX's IP address.
- 6. Select **Validating**. The login information is checked, a dialog is opened which displays the result of the check. Use **OK** to close the window.
- 7. If in the PBX the option Automatic exchange line request is deactivated: Activate in the printer driver the checkbox Preset automatically to put the exchange line access number in front of the recipient's fax number.

- 8. If in the PBX the option **Automatic exchange line request** is deactivated: In the entry field **Exchange line access number** enter the exchange line access number in the PBX.
- 9. If necessary, select **New** and repeat steps 2 to 8 to configure another fax box of the PBX in the printer driver.
- 10. Select **OK** and **Close**.

Send Fax to a Recipient

Requirements:

- At least one reserved channel for outgoing calls of the voice mail/fax function.
 - Administration > VoIP > VoIP configuration
- · Configured fax box
- Installed and configured printer driverCOMpact/COMmander Fax
- 1. Open the document you wish to send as a fax.
- 2. Print the file and select in the printer selection dialogue **COMpact/ COMmander Fax**.
- 3. Enter under **Recipient's fax number** the corresponding phone number (special characters: + () () (-).
 - Important: Depending on the fax box configuration in the printer driver, the exchange line number in the PBX is automatically put before the phone number or it has to be entered before the Recipient's fax number.
 - Note: The entered phone number is automatically converted into a character chain before sending.
 - **Note:** If a phone number has been saved to the clipboard, it is automatically transferred to the entry field.
- 4. Select under **Fax box for sending faxes** the desired fax box.
- 5. Select **Send** and then **Close**.
 - Note: If you want to cancel the fax transmission, you can delete the fax message in the web interface of the PBX under **User data** > **Messages**.

Send Fax to several Recipients

Requirements:

- At least one reserved channel for outgoing calls of the voice mail/fax function.
 - Administration > VoIP > VoIP configuration
- · Configured fax box
- Installed and configured printer driverCOMpact/COMmander Fax
- Create a text file (*.csv, *.txt) with the phone numbers of the fax recipients (max. 18 digits per phone number, special characters: + ()
 / , :)
 - Important: Depending on the fax box configuration in the printer driver, the exchange line number in the PBX is automatically put before the phone number or it has to be entered before the fax number of the recipient.
- 2. Open the document that should be sent as fax.
- 3. Print the file and select in the printer selection dialogue **COMpact/ COMmander Fax**.
 - **Note:** Entries under **Recipient's fax number** are not considered in case of a broadcast file.
- 4. Select under **Fax box for sending faxes** the desired fax box.
- 5. Select Broadcast file.
- 6. Select the in step 1 created text file and select **Open**.
 - Note: If the error message 507 occurs, fax transmission is interrupted for 3 minutes and then restarted. After 10 unsuccessful attempts the transmission is stopped.

7. Select Close.

Note: If you want to cancel the fax transmission, you can delete the fax message in the web interface of the PBX under **User data** > **Messages**.

VoIP

VoIP

VoIP channels
External Internet Telephony
Internal IP telephony
External Private Branch Exchanges

VolP channels

The number of available VoIP channels depends on the codecs used, defined by the set VoIP operation mode. The overviews below give information about possible combinations.

COMtrexx Business:Installed and configured 4FXS modules reduce the available VoIP operation modes and therefore also the maximum number of available VoIP channels. If a 4FXS module has been added at a later point of time so that the current VoIP operating mode is not possible, the VoIP operation mode has to be changed manually.



Further help under <u>Hardware</u>

COMtrexx VM: overview of the available channels and codecs

VoIP operating mode		Classic	VoIP oriented	VoIP maximum
VoIP channels (max. possible)		190	370	500
codecs internal	G.711	+	+	+
	G.729	+	-	-
	G.722	+	_	_
codecs external	G.711	+	+	+
	G.729	+	+	_
	G.722	+	-	-

COMtrexx Business: overview of available channels, codecs and modules

VoIP operating mode		Classic	Broadban	VoIP oriented	VoIP maximum
VoIP channels (max. possible)		54	64	66	80
codecs internal	G.711	+	+	+	+
	G.729	+	-	-	-
	G.722	+	+	-	-
codecs external	G.711	+	+	+	+
	G.729	+	-	+	-
	G.722	+	+	-	-
a 4FXS module		+	+	+	-
two to five 4FXS modules		+	_	-	-
NET module		+	+	+	+

Configuration

• Configuring VoIP operation mode.

VoIP channels

- Administration > VoIP > VoIP configuration > Presetting of the VoIP channels
- · Distributing the VoIP channels.
 - Administration > VoIP > VoIP configuration > Distribution of the VoIP channels
- · Reserving VoIP channels for VoIP provider.
 - Exchange lines > Providers and Accounts > / > RTP

External Internet Telephony

Using Internet telephony, calls and data are transferred via the Internet. This requires an account at a VoIP provider. The transmission quality and the reliability of Voice over IP depends to a great extent on the quality of the Internet connection used.

Configuration of a DNS server is necessary for translating the phone number.



Further help under Network

Configuration

- Configuring at least one VoIP channel as external VoIP channel.
 - Administration > VoIP > VoIP configuration > Distribution of the VoIP channels
- Creating a VoIP account.
 - Exchange lines > Providers and Accounts > + New
- If required, importing current VoIP provider data.
 - Exchange lines > Providers and Accounts > ↓ Import
- Entering account and access data for the VoIP account.
 - Exchange lines > Providers and Accounts > / > Access data
- · Checking registration.
 - Exchange lines > Providers and Accounts > Status
- Creating a call distribution for incoming calls.
 - Exchange lines > Exchange line routing > Exchange line routing incoming > Call distribution for

- If required, making VoIP settings (jitterbuffer, echo compensation) for increasing the VoIP call quality.
 - Exchange lines > Providers and Accounts > / > RTP
- If required, reducing the Interval NAT Keep Alive.
 - Note: If no calls are possible via VoIP, although account registration and perhaps even an initial call were successful, this might have something to do with a very short timeout configured in the firewall. Reduce the Interval for NAT Keep-Alive for all used providers.
 - Exchange lines > Providers and Accounts > / > SIP

Internal IP telephony

The connected VoIP telephones receive the data needed for registration (VoIP registrar, internal number and password) by provisioning via the provisioning server of the PBX. Before configuring internal numbers and associated passwords, users have to be created.

- Further help under <u>User</u>
- Further help under Provisioning

By upgrading with a NET module (only COMtrexx Business), there is the possibility to operate the internal IP telephony in a network separated from an external Internet telephony.

Further help under Network interface (second)

Configuration

- Configuring a VoIP channel as internal channel for each internal VoIP user planned.
 - Administration > VoIP > VoIP configuration > Distribution of the VoIP channels
- If required, configuring the SIP port of the internal VoIP registrar.
 - Administration > VoIP > VoIP configuration > SIP
- If required, switching on the DiffServ to increase the VoIP call quality.
 - Administration > VoIP > VoIP configuration > Quality of Service (QoS)
- If required, changing the jitterbuffer value to increase the VoIP call quality.

Telephones > Assignment > /

External Private Branch Exchanges

The PBX handles external extensions in the same way as internal VoIP telephones.

Further help under Internal IP telephony

VoIP telephones are connected to the PBX network via a VPN tunnel.

- Note: If necessary, contact your network administrator to find out the VPN settings.
 - Further help under VPN
- Important: When configuring an external extension, remember that these telephones can only make emergency calls if specific requirements have been met.
 - Further help under **Emergency calls**
- Important: When setting up an external extension, please note that the availability of an NTP server must also be guaranteed for these telephones.
 - Further help under Date and Time

VPN

In order to be able to securely connect remote devices to the PBX, the PBX offers an integrated OpenVPN server.

- Important: The OpenVPN server is only available with feature level 2 or higher. If you want to use this function, please contact your dealer or system partner.
 - Further help under Feature level

Configuration

- · Configuring and enabling OpenVPN Server.
 - Administration > Network > VPN
 - An example configuration for the connection of COMfortel D-series telephones can be found in the Auerswald DokuWiki.

Using the web interface

Basic Knowledge

Dashboard

Hardware

Exchange lines

User

Groups

Voice mail and fax boxes

Telephones

Internal Devices

External Devices

Functions

User data

Administration

Monitoring and Overviews

Basic Knowledge

Requirements and Notes
Open the Web Interface
Operating Instructions

Requirements and Notes

The web interface of the PBX is opened with a browser and allows the subdivision into two permission levels:

- Administrator (admin)
- User

Requirements:

- Recommended browsers: Mozilla Firefox, Google Chrome (current version)
- You need to know the PBX's IP address.
- · Configured password

Every login is logged (page **Administration** > **Service** > **Logging**). Additionally this information can be printed (page **Administration** > **Maintenance** > **System messages**).

After the third failed login attempt, for a minute it is not possible to log in. This blocking period will be increased after another three failed login attempts.

If logged in as user, only parts of the web interface are displayed.

Open the Web Interface

As an administrator

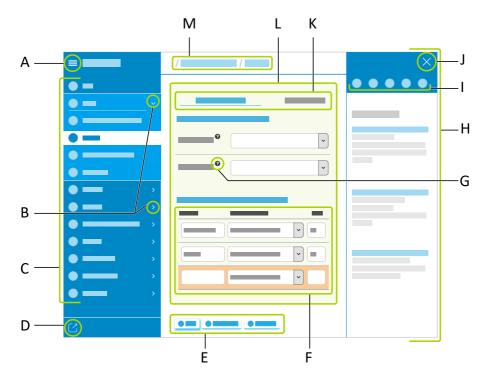
As an administrator

- 1. Enter the IP address of the PBX (for example, 192.168.0.240).
- 2. Enter the admin user name and the corresponding password.

Operating Instructions

Overview
Fields and User Elements
Display elements

Overview



- A Displays or hides the vertical navigation
- B Displays or hides submenus
- C Vertical navigationHere, click on the required topic to select it.
- D Logs out the user
- E Interactive buttons such as for saving, deleting, exporting made settings

The buttons differ according to the area.

- F List to which further settings can be added
- G Context-sensitive help

Opens the integrated help on the right screen margin for detailled information about a setting.

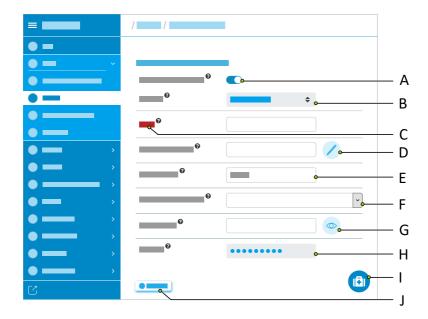
- H Display of the contents of the integrated help
- I Navigation elements within the integrated help, from left:
 - Menu
 - Search
 - Back
 - Next
 - Home screen
- J Hides the integrated help
- K Tabs
- L Content area

Displays the available settings for the desired area.

M Breadcrumb navigation

Displays the branching within the web interface.

Fields and User Elements



A Slide toggle

Enables or disables settings and functions.

- B Pre-filled selection field
- C Required entry field
- D Edit further properties

Opens an overlaying menu for further settings.

- E Text input field
- F Selection field

Shows a selection of possible settings.

G Shows detailled settings

Fields and User Elements

- H Hidden password entry
- I Displays the integrated help on the right screen margin
- J Saves all settings

Display elements





Display How many are assigned/total or How many are unrestricted

Meaning depends on context: for example, memory use, phone number assignments, language templates, audio files etc.



Displays notifications.



Displays notes.



Displays warnings.



Hides notifications, notes or warnings.

The appropriate icon is displayed on the upper right-hand edge of the screen until the error has been resolved or the message has been temporarily closed.

Dashboard

Status information Language Feature level

Status information

Type

Displays the used device type.

Firmware version

Displays the firmware version currently in use on your device.

Further help under <u>Firmware Update</u>

Serial number

Displays your device's serial number.

MAC address

Displays the specific and unique hardware address of the PBX.

Clipboard

Copies status information to the clipboard, so you can access it easily, for example, to make a request for support.

Online Help

Displays the online help in your browser.

Legal

Refers to the documents **Data protection**, **End User License Agreement for Software (EULA)** and **Open source licenses**.

Data protection

Displays a multi-lingual PDF that contains information about the way Auerswald processes data, and how data is sent to Auerswald.

End User License Agreement for Software (EULA)

Displays a multi-lingual PDF that contains Auerswald's EULA.

Open source licenses

Displays a multi-lingual PDF that contains information about the open source licenses that are in use.

Language

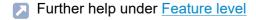
Language web interface

Changes the language of the web interface.

Feature level

Current

Currently activated feature level for this PBX.



Maximum

Maximum possible feature level.

Further help under Feature level

Hardware

Slot

Slot in the casing of the COMtrexx Business.

Further help under <u>Hardware</u>

Configuration

Module on this slot assigned to the configuration, also preparing. When operating, the selection must match the hardware found.

not in use

No module is inserted or the inserted module is not to be used in the configuration.

Further help under <u>Hardware</u>

Hardware found

Module currently inserted into this slot.

Further help under Hardware

Exchange lines

Exchange line request/locations
Emergency calls
Providers and Accounts
Exchange line routing

Exchange line request/locations

Exchange line request / Exchange line access number Miscellaneous Locations

Exchange line request / Exchange line access number

Automatic exchange line request

The PBX automatically distinguishes between internal and external numbers with the analysis of the digit sequence and the adjustment of the dialled phone number and the internal dial plan. Therefore, dialling an exchange line access number is not necessary for external phone numbers.

- Further help under Internal numbers
- Further help under Exchange line request

Exchange line access number

Is required to request an external line. The exchange line access number is only effective when the automatic exchange line request is disabled. Changing the preset 0 results in a limitation of the dial plan and is only meaningful if another exchange line access number is normally used (e.g. in England).

- Further help under Internal numbers
- Further help under <u>Exchange line request</u>

Miscellaneous

Tones

The standard depends on the configured country. For these tones, only the frequencies 400, 425, 440 and 450 Hz are possible. Tones differing from these frequencies are rounded to the nearest frequency. Changes should only take place on order of the manufacturer support.

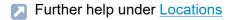


Further help under Tones

Locations

Name

Freely selectable name. This name is used to identify the location within the PBX.



Country

When you configure the country the PBX is automatically assigned the default settings that are required in the selected country (e.g. sounds, emergency numbers). If the country required is not listed, you can enter the according country code by selecting **International**.

Further help under Locations

Country code

International phone prefix (e.g. 0044 for the UK). These are necessary for some functions of the PBX (e.g. Call Through, block- and allowlists).

Further help under Locations

Local area code

Phone number prefix (e.g. 0208 for Greater London). These are necessary for some functions of the PBX (e.g. Call Through, block- and allowlists). Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their local area codes.

Further help under Locations

Dial level

The selection of the dial level (Location, National, International) is country-specific. Depending on the country and the settings for the local area code, a dial level must be selected (example: In Belgium, local area codes do not exist, therefore the dial level area should not be selected).

Further help under Locations

Use for

Displays which accounts the location has been assigned to.

Further help under Locations

+ New

Creates a new entry.

Further help under Locations

m Delete

Deletes the entries marked in the list.

Emergency calls

Emergency call has priority
Emergency and service numbers

Emergency call has priority

Emergency call has priority

An external line is definitely made available from the PBX when dialling the emergency number. If necessary, a call can be interrupted to enable an emergency call.

- Important: If this function is disabled and other calls are being made during an emergency, there is a risk that the emergency call cannot be made immediately.
- Further help under **Emergency calls**

Emergency call only with exchange line access number

Emergency numbers must be dialled with the exchange line access number first (e.g. 0112 instead of only 112). Therefore, the internal dial plan is not reduced by special numbers.

- Important: If the function is enabled, there is a risk that the exchange line access number is forgotten in an emergency so that no emergency call can be made.
- Further help under <u>Emergency calls</u>

Emergency and service numbers

Legal emergency numbers
User-defined emergency numbers
Service numbers

Legal emergency numbers

Phone number

Special numbers that do not follow the standard numbering scheme without local area code, which reduces the size of the internal dial plan.

- Further help under Internal numbers
- Further help under **Emergency calls**

Name

Freely selectable name. This name is used to identify the emergency or service numbers within the PBX.

Further help under **Emergency calls**

Country

Indicates in which country the phone number can be used.

Further help under **Emergency calls**

+ New

Creates a new entry.

Further help under Emergency calls

∄ Delete

Deletes the entries marked in the list.

User-defined emergency numbers

Phone number

Phone numbers that follow the standard numbering scheme with local area code.

Some countries (e.g. Denmark, Greece, Luxembourg) do not have a fixed number sequence for their local area codes. If this also applies to the location of the PBX, you do not need to enter a local area code.

Further help under **Emergency calls**

Name

Freely selectable name. This name is used to identify the emergency or service numbers within the PBX.

Further help under <u>Emergency calls</u>

Country

Indicates in which country the phone number can be used.

Further help under <u>Emergency calls</u>

+ New

Creates a new entry.

Further help under **Emergency calls**



Deletes the entries marked in the list.

Service numbers

Phone number

Special numbers that do not follow the standard numbering scheme without local area code, which reduces the size of the internal dial plan.

- Further help under Internal numbers
- Further help under **Emergency calls**

Name

Freely selectable name. This name is used to identify the emergency or service numbers within the PBX.

Further help under Emergency calls

Country

Indicates in which country the phone number can be used.

Further help under **Emergency calls**

+ New

Creates a new entry.

Further help under Emergency calls

m Delete

Deletes the entries marked in the list.

Providers and Accounts

Overview Account Provider

Overview

Provider name

Provider with whom the account was set up.

Further help under External Internet Telephony

Account name

Freely selectable name. This name is used to identify the account within the PBX.

Further help under External Internet Telephony

Connection type

Connection type to be used with the account.

Further help under External Internet Telephony

Use

The PBX uses the account for external calls.

Further help under External Internet Telephony

Status

Displays the actual status of the VoIP account.

Further help under External Internet Telephony

Name of the provider | / Configure

Opens the configuration of the provider.

- Note: You can obtain the data for the settings either directly from the provider or from suitable lists in the Internet.
- Details about settings under <u>Provider</u>

Name of the account | / Configure

Opens the configuration of the account.

- Note: You will receive the settings data from the provider with whom you have set up the account.
- Details about settings under <u>Account</u>

+ New

Creates a new entry for the account list.

Adding

If the required provider does not exist, it can be added by importing a file or loading of a provider template from the Internet.

Further help under External Internet Telephony

Delete

Deletes the entries marked in the list.

丛 Import

If the required provider does not exist, it can be added by importing a file or loading of a provider template from the Internet.

Further help under <u>Templates</u>

Account

Settings

Phone numbers(for Point-to-Multipoint Connection)
Phone numbers (for PBX connection)
Expert configuration

Settings

Network Provider
Access data
Miscellaneous
Phone numbers (for PBX connection)

Network Provider

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Account has emergency call ability

This account allows emergency calls to be made, provided that the provider has authorised and switched on emergency call ability.

- Note: When switched off, this account cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This telephone does not allow emergency calls.

 Please use an alternative."
- Note: The phone allows emergency calls by default. Check if you can actually use your account to make emergency calls. Not all network providers allow emergency calls.
- Further help under <u>Emergency calls</u>

Location

Location for which the provider provides the account. The correctly assigned location is necessary, among other things, so that the PBX can properly initiate an emergency call.

Further help under <u>Locations</u>

Access data

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Exchange line access number (account number)

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This is then used for the specific VoIP access point.

- Further help under Internal numbers
- Further help under External Internet Telephony

User name

User name assigned by the provider when the account was set up. It is used to identify the user to the provider.

- Note: Note that some providers allocate both login information for the web page and separate login information for using VoIP (e.g. a dialin password). Find out what information is needed for the dial-in and enter it accordingly.
- Note: The user name is called differently by different providers. For example, the terms Username, Authorization User, SIP code/SIP password and SIP-ID are all used to describe a user name.
- Further help under External Internet Telephony

Password

Password assigned by the provider when the account was set up. It is used to verify the user name.

- Note: Note that some providers allocate both login information for the web page and separate login information for using VoIP (e.g. a dialin password). Find out what information is needed for the dial-in and enter it accordingly.
- Note: The password is called differently by different providers. For example, it can also be called a customer password.
- Further help under External Internet Telephony

Authorisation ID

Authorisation ID (authentication ID) assigned by the provider for the account.

- Note: You can only input an entry in this field if the provider has assigned an authorisation ID.
- Further help under External Internet Telephony

Miscellaneous

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

CLIP no screening

The customer-specific phone number (e.g. a service phone number) entered in the exchange line routing is sent to the call destination without checking whether this number is correct.

- Note: CLIP no screening must be supported by the used privider. Additionally, a number presentation type must be configured for the provider: In accordance with RFC 3325 with P-Asserted-Identity or In accordance with RFC 3325 with P-Preferred-Identity.
- Note: When an emergency number is dialled, the phone number received from the network provider is always transmitted, not the phone number you specified yourself.
- Note: However, the correct phone number is also still transferred to the exchange line, but this cannot be viewed by the person being called.
- Further help under Number presentation

Expert configuration

Opens the expert configuration of the account.

Miscellaneous

Details about settings under <u>Expert configuration</u>

Phone numbers (for PBX connection)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

PBX base number

The PBX base number assigned to the account.

Further help under External Internet Telephony

Extensions block

The Direct Dial In assigned to the account.

Further help under External Internet Telephony

Number presentation

The phone number (entry of a DDI from the extensions block) to be presented for this account.

Further help under Number presentation

Phone numbers(for Point-to-Multipoint Connection)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Phone numbers

The phone number assigned to the account.

Further help under External Internet Telephony

Display name

Freely selectable name.

- Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under Display name.
- Further help under External Internet Telephony

Number presentation

The phone number to be presented for this account.

Further help under Number presentation

+ New

Creates a new entry.

Further help under <u>External Internet Telephony</u>

Phone numbers(for Point-to-Multipoint Connection)



Deletes the entries marked in the list.

Phone numbers (for PBX connection)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Additional numbers

Phone numbers different from the extensions block.

Further help under External Internet Telephony

Display name

Freely selectable name.

- Note: If your provider (in Germany) is T-Online, you must also enter the VoIP phone number you received from them under Display name.
- Further help under External Internet Telephony

+ New

Creates a new entry.

Further help under External Internet Telephony

iii Delete

Deletes the entries marked in the list.

Expert configuration

VoIP account
Call forwarding general
Call forwarding unconditional (CFU)
Call Forwarding if Busy (CFB)
Call forwarding no reply (CFNR)

VoIP account

Account

Select the account to be displayed/configured.



Further help under External Internet Telephony

Copy settings to

Copies the expert configuration settings to the selected account.

Call forwarding general

Account

Select the account to be displayed/configured.



Further help under External Internet Telephony

Call forwarding method

Call forwardings at the exchange line are configured and performed in different ways. The exchange line, router or the provider determines which one of the call forwardings listed below can be used.

Keypad

The call forwarding is controlled in the PTMP exchange line. The digit sequence depends on the provider or router.

Call Deflection

Call forwarding shall be executed directly by the exchange line. The PBX sends a corresponding message with new destination number to the exchange line. Accordingly, the call is forwarded to the new destination by the provider/exchange line.

2nd B channel

The call forwarding is performed by the PBX during a call. It is possible for all exchange line types but requires a free voice channel and the number of the caller cannot be displayed unless CLIP no screening is used which is not supported by many providers.

Further help under Call forwarding

For call forwarding via the PBX, use any available call channel

When an external number is forwarded, the call can be forwarded via a call channel on any of the other accounts.

- Note: When switched off, a call can only be redirected via a call channel on the same account. The probability that the call cannot be forwarded because the channels are busy is therefore higher.
- Further help under Call forwarding

Renew Call Forwarding at PBX reboot

When rebooting the PBX, the saved data of the Call Forwarding is renewed.

- Important: In Switzerland, the configuration of a call forwarding is chargeable. To avoid costs, the check box must be disabled.
- Further help under Call forwarding

Delay time for call forwarding on no reply via PBX

This delay time for external numbers specifies the time (in seconds) after which a call is forwarded if the called user does not respond.

- Note: The waiting time applies only to call forwarding that was set up using the PBX. If the network provider has configured the call forwarding function, the waiting time depends on their settings and cannot be configured in the PBX.
- Further help under Call forwarding

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions call forwardings are performed.



Further help under Exchange line routing

Call forwarding unconditional (CFU)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Call forwarding available in exchange line

If the exchange line has a CFU, this function can be enabled.

If the exchange line does not have a CFU, this function must be disabled.

Further help under <u>Call forwarding</u>

Keypad/Registro code: Enabling CFU

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- · Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under Call forwarding

Keypad/registro code: Disabling CFU

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under <u>Call forwarding</u>

Call Forwarding if Busy (CFB)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Call forwarding available in exchange line

If the exchange line has a CFB, this function can be enabled.

If the exchange line does not have a CFB, this function must be disabled.

Further help under <u>Call forwarding</u>

Kaypad/Registro code: Enabling CFB

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under Call forwarding

Keypad/Registro code: Disabling CFB

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under <u>Call forwarding</u>

Call forwarding no reply (CFNR)

Account

Select the account to be displayed/configured.

Further help under External Internet Telephony

Call forwarding available in exchange line

If the exchange line has a CFNR, this function can be enabled.

If the exchange line does not have a CFNR, this function must be disabled.

Further help under Call forwarding

Keypad/Registro code: Enabling CFNR

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under Call forwarding

Keypad/Registro code: Disabling CFNR

Not at PBX connection

The feature codes specify the character strings the VoIP network provider requires to execute a service feature on the VoIP connection.

- Digits:0-9
- Initials* and #
- Initials, (comma) for a 0,5 second pause
- The **n** as a wildcard for the destination number (CDPN)
- The **m** as a wildcard for the own phone number (MSN)
- Further help under <u>Call forwarding</u>

Provider

Provider

SIP

RTP

Settings

Number presentation (outgoing)

Evaluation of incoming SIP messages

Provider

Provider

Select the provider to be displayed/configured.

Further help under External Internet Telephony

Name

Freely selectable name. This name is used to identify the provider within the PBX

Further help under External Internet Telephony

∴ Export

The displayed provider is saved to the data storage device (PC) in an * xml file

Further help under External Internet Telephony

- Copy

Uses an existing provider as the basis for creating a new provider.

Further help under External Internet Telephony

SIP

Provider

Select the provider to be displayed/configured.



Further help under External Internet Telephony

Domain

The domain (also known as a "realm", "SIP domain" or "SIP realm") is required for the VoIP address. The structure of a user's VoIP address is similar to an e-mail address. It is made up of the VoIP phone number and the domain, separated by the @ character: <user>@domain.



Further help under External Internet Telephony

Name resolution

There are both country-specific and provider-specific reasons why name resolution is not always successful via the default gateway. In such cases, another way should be chosen here.

Name resolution via IP configuration

There are several accounts at this domain and the name resolution for another account at this domain has already been configured.

Note: It is possible to set the name resolution differently for the accounts of the same domain.

via used Internet access

The name resolution takes place through the IAD selected as used Internet access. The DNS server provided by the provider is used.

Use own DNS (via IAD)

The DNS servers that are entered below are used via the selected IAD as Internet access.

Further help under External Internet Telephony

First DNS server / Second DNS server

IPv4 or IPv6 address. In case of an IPv6 address, the IAD chosen as used Internet access must also have an IPv6 address.

Further help under External Internet Telephony

Used Internet access

Select the Internet access for this provider. A default IAD with the gateway of the first network interface is automatically created. If you only want to use the first network interface, you can select the default IAD. If using both network interfaces, if necessary, the IAD to be selected must be configured first.

Further help under External Internet Telephony

Registrar

REGISTER requests are sent to the specified registrar (also known as a SIP registrar, registry, SIP server, SIP registry server).

Note: The registrar should only be disabled if the provider requires this. No NAT Keep-Alive is performed if the registrar is disabled. An internal network is usually protected from actions from the outside by, for example, a firewall integrated in the router. Without NAT Keep-Alive, the firewall's security functions will defend you against attempted accesses from the Internet. For this reason, you may need

to implement port forwarding for incoming SIP packets on the SIP port set up in the provider's configuration.

Further help under <u>External Internet Telephony</u>

Registrar | Address

Registrar's IP address or URL.

Further help under External Internet Telephony

Registrar | Port

Port of the registrar. The standard value is 5060 (recommended).

- ▲ Caution: Divergent entries may lead to malfunctions as soon as the provider makes changes. Only the standard value allows an SRV query.
- ▲ Caution: Each time a port is opened on the router, this creates a potential security risk. It is essential you put additional protective measures in place.
- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Further help under External Internet Telephony

Registrar | Time lapse for the registration

The registration time specifies how many minutes the PBX waits before reregistering itself with the provider.

Note: In the test phase, after configuring the provider, we recommend you enter a low value (e.g. three minutes). After completing the test

phase, you can enter a much higher value, since some providers also reject calls if too many attempts are made to register.

Further help under External Internet Telephony

Registrar | SIP registration in accordance with RFC 6140

The SIP protocol SIPconnect 1.1 is used by the PBX.

Further help under External Internet Telephony

Registrar | Re-Registration after SIP Forbidden (403)

If a SIP Forbidden (403) is received on an INVITE, the involved provider is re-registered.

Further help under External Internet Telephony

Registrar | Re-Registration after SIP Timeout (408, 503)

If a SIP Timeout (408, 503) is received on an INVITE, the involved provider is re-registered.

Further help under External Internet Telephony

NAT traversal

If NAT traversal is switched on and a query is sent from a local IP address to the public network, the sending IP address is swapped with the public IP address. This function is performed in the reverse direction for the reply.

deactivated

NAT traversal is not performed by the PBX, but the local address is used. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

activated (STUN)

NAT traversal is performed by the PBX. The STUN server entered for the provider is used. Only in connection with IPv4.

activated (rport)

NAT traversal is performed by the PBX. To achieve this, the registrar for the provider must be switched on. The SIP request additionally contains the part rport, in which the IP addresses used (local, public) are transported.

- **Note:** If problems occur with single-sided call connections, use the option with use of STUN instead.
- Caution: Each time a port is opened on the NAT router, this creates a potential security risk. It is essential you put additional protective measures in place.

Public IP

If the provider requires it: NAT traversal is performed by the PBX via a static public IP address.

use static public IP address in VIA Header

If the provider requires it: The static IP address is used in the VIA Header.

use static public IP address in Contact Header

If the provider requires it: The static IP address is used in the Contact Header.

Further help under External Internet Telephony

NAT Keep-Alive | active

NAT Keep Alive data packets are sent according to the selected method and the set interval.

Further help under External Internet Telephony

NAT Keep-Alive | Method

Select the data packets sent to maintain the NAT mapping.

Default

Sending NAT Keep Alive data packets

SIP OPTIONS

Sending SIP OPTIONS data packets

Further help under External Internet Telephony

NAT Keep-Alive | Interval

After the time specified here has passed, NAT Keep Alive data packets are sent to the Firewall to maintain the NAT mapping.

- Note: Some providers lock accounts if the NAT keep alive interval is too short. This is usually reported with error message 503 during SIP registration. If this problem occurs, it is recommended to reduce the value (e.g. to 180).
- Further help under External Internet Telephony

Outbound Proxy/Session Border Controller

deactivated

The PBX does not use outbound proxy.

automatically

The PBX automatically determines an outbound proxy.

manually

The PBX uses the outbound proxy configured for the provider.

- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Further help under <u>External Internet Telephony</u>

SIP port

The SIP port on the PBX is used both as an incoming port and an outgoing port (i.e. for communications with the provider). The SIP port must be different for each provider and cannot be the same as the SIP port of the internal VoIP registrar or the SIPS port of the internal VoIP registrar.

- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Further help under External Internet Telephony

SIP session timer

At the specified intervals, the SIP session timer checks whether the connection is still present. (The PBX cannot detect whether a call connection is disrupted due to problems on the provider side.)

Note: When the SIP session timer is switched on, this can result in increased breaks in conversations after the set interval, if a provider

has not implemented the renewal of the session cleanly. In this case, you should adjust the interval of the SIP session timer or switch it off.



Only en-bloc dialling

A connection is not set up to transmit the digits dialled here to the provider until the delay time after the last digit is dialled has passed, all the digits are then sent as a block. Time measurement restarts after the entry of each digit. After the selected time has passed, the PBX recognises the most recent digit as the final digit and will not allow any more numbers to be entered. Select as short a time as possible, because the PBX does not forward the entire phone number to the central office until the specified time has passed. Approximately three seconds is the default value for manual inputs.

Further help under External Internet Telephony

SIPS

This provider encrypts external connections to prevent listening in. The processes used to set up and shut down a connection, and signalling, are all encrypted by SIPS.

Further help under <u>External Internet Telephony</u>

SIPS | URL structure

SIPS

Encryption of SIPS via TLS.

SIP/TLS

Encryption of SIP via TLS. SRTP is used to encrypt call data.

Further help under External Internet Telephony

SIPS | Disable host name check

No verification whether the certificate belongs to the domain/IP.

Further help under Certificates

SIPS | Manage certificates

Displays information about the stored certificate or opens the certificates management.

Further help under Certificates

SIPS | Certificate

Certificate for one-way SSL authentication.

Further help under <u>Certificates</u>

SIPS | Client certificate

Client certificate (.pem format) for the mutual SSL authentication.

Further help under Certificates

SIPS | Client key

Client key (.pem format) for the mutual SSL authentication.

Further help under Certificates

SIPS | Client passphrase

Passphrase that may belong to the private key.

Further help under Certificates

SIP transport

Transport protocol used.

Further help under External Internet Telephony

IP protocol

IPv4

IPv4 operating mode is used for communications with the provider.

IPv₆

IPv6 operating mode is used for communications with the provider.

Automatically

The PBX first checks whether the provider supports IPv6. If so, operating mode IPv6 is used for communications with the provider. Otherwise, operating mode IPv4 is used.

- Note: If the actual IP operating mode is IPv6, STUN cannot be used as a NAT method for neither SIP nor RTP.
- Further help under <u>External Internet Telephony</u>

RTP

Provider

Select the provider to be displayed/configured.

Further help under External Internet Telephony

Maximum number of VoIP channels provided by the provider

Number of VoIP channels provided by the provider. If these channels are occupied, additional connections required are made using another provider.

Further help under External Internet Telephony

VoIP channels reserved for incoming calls

Number of VoIP channels provided by the provider to be used only for incoming calls.

Further help under External Internet Telephony

VoIP channels reserved for outgoing calls

Number of VoIP channels provided by the provider to be used only for outgoing calls.

Further help under External Internet Telephony

SRTP

deactivated

Forces deactivation of call data encryption via SRTP. The connection is not established if the call partner (provider, other PBX in sub-system operation or external VoIP telephone) requires call data to be encrypted using SRTP.

preferred

Switches on negotiation for the encryption of call data via SRTP. When a call is made, the call partner will be asked whether encryption is possible. If so, the call data will be encrypted and then transmitted. If not, the call data will not be encrypted.

required

Forces activation of call data encryption via SRTP. The connection is not established if the call partner (provider, other PBX in sub-system operation or external VoIP telephone) does not require call data to be encrypted using SRTP.

- Note: SIPS should also be enabled at the same time, otherwise the key for SRTP encryption would be readable.
- Further help under <u>External Internet Telephony</u>

NAT traversal

If NAT traversal is switched on and a query is sent from a local IP address to the public network, the sending IP address is swapped with the public IP address. This function is performed in the reverse direction for the reply.

deactivated

NAT traversal is not performed by the PBX, but the local address is used. The router which connects the local area network with the Internet should be a properly functioning SIP-aware router which performs NAT traversal.

activated / STUN

NAT traversal is performed by the PBX. The STUN server entered for the provider is used. Only in connection with IPv4.

Permanent IP

If the provider requires it: NAT traversal is performed by the PBX via a static public IP address.



DTMF signalling

Inband

The PBX uses the same channel to transfer the DTMF signals and the voice data (DTMF tones). Requirement for Inband signalling is an uncompressed codec (G.711).

Outband (RFC 2833)

The PBX uses different channels to transfer the DTMF signals and the voice data. The DTMF tones are filtered out of the voice data.

both procedures

The PBX transmits the DTMF signals on two channels (1st contains DTMF tones together with the voice data + 2nd contains the DTMF signal).



Echo cancellation

Local echoes and reverberation effects are comepnsated.

Further help under External Internet Telephony

Automatic silence suppression

(Voice Activity Detection)

This function allows to add the header silenceSupp: on/off to the SDP negotiation of the SIP packets.

Automatically

The header is not set (recommended).

On

Use this option only tentatively for known problems with silent suppression.

Off

Use this option only tentatively for known problems with silent suppression.



Further help under **External Internet Telephony**

Comfort Noise Support

Automatically

(recommended)

On

Comfort noise is offered during SDP.

Off

Comfort noise is not offered during SDP. This option is only needed in case of compatibility problems with comfort noise.



Further help under External Internet Telephony

Jitter buffer

The size of the jitter buffer specifies how many RTP packets can be cached, to buffer disruptions or compensate for them.



Further help under External Internet Telephony

maxptime

Specifies the maximum size of a received packet for RTP supported by the PBX (30 ms recommended).

suppress

Suppresses the output of the maxptime header.

Automatically

The PBX attempts to determine the required value itself.



Further help under External Internet Telephony

Use the exchange line replacement tone

The PBX generates the exchange line replacement tone in the case, for example, of longer dialling phases during which the PBX has not yet sent a call tone. If you find this disruptive or annoying, it can be switched off.

Automatically

The PBX generates the exchange line replacement tone in the case of longer dialling phases during which there is no call tone.

Off

No exchange line replacement tone is generated. No acoustic acknowledgement during longer dialling phases during which there is no call tone



Further help under External Internet Telephony

Codec settings

The PBX makes various codecs available. The selection of a codec affects the quality of a VoIP call. Different codecs can be configured, depending on the available connection bandwidth, ranging from codecs with the best possible VoIP call quality down to codecs with high compression (low bandwidth). The codec actually used for call handling is defined by the codec negotiation with the provider.

Quality

Selects a codec sequence with best possible VoIP call quality (high bandwidth) as highest priority.

Compromise

Selects a compromise between VoIP call quality (high bandwidth) and compression (lower bandwidth) for the codec sequence.

Compression

Selects a codec sequence with stronger compression (lower bandwidth) as highest priority.

- Note: The preset codec sequence can be changed manually if necessary.
- Note: If there are disruptions regarding the call quality when using codecs with high bandwidth (e.g. G.711), the bandwidth of the connection may not be sufficient. If the call quality is impacted

RTP

frequently, it is a good idea to only select codecs with lower bandwidths in the Codec settings priority list.

Further help under External Internet Telephony

Settings

Provider

Select the provider to be displayed/configured.

Further help under External Internet Telephony

STUN server (IP address or URL)

STUN server set for the provider for use of NAT traversal.

Further help under External Internet Telephony

STUN server | Port

Port of the STUN server.

- Note: If you encounter problems with unilateral call connections, enabling the RTP port might help.
- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Caution: Every time port forwarding is performed, there is a security risk. For this reason, we recommend you use port forwarding as little as possible.
- Further help under External Internet Telephony

STUN server | Interval for STUN server query

Determines after what time STUN server reuqests are sent.

Further help under <u>External Internet Telephony</u>

Audio is switched through at call forwarding via 2nd B-channel

The connection between the caller and the PBX is always created when a call is forwarded externally. As a result, the caller will hear all replayed elements (tones, announcements) which are played by the provider when the connection with the call forwarding destination is established.

- Note: If external call forwarding is configured over a provider, the call tone may not be replayed for the caller (the connection to the call forwarding destination number will therefore be switched through in a "surprising" way).
- Further help under External Internet Telephony

Switch audio on call start

Audio is switched through to the provider provided that the provider supports Early Media. This is useful if, for example, a callback request has to be confirmed with "Yes".

Further help under <u>External Internet Telephony</u>

EarlyMedia Support

During the call phase, audio such as call tone, signal tone, announcement or language can be transmitted (EarlyMedia). The following options define the use of **EarlyMedia** (RFC 3261) in general and the use of the SIP header **P-Early Media** (RFC 5009).

If EarlyMedia or the SIP header P-Early Media is supported, is up to the provider's discretion. It is possible that the provider allows the negotiation but suppresses the audio transmission.

The following options allow to adjust the PBX's reaction to the reaction of the provider.

Disabled

No audio is transmitted during the call phase. But the status of the call is signalled and the end device of the caller then generates for example the call tone or the busy tone.

Announcements are still suppressed. No tones are generated for the caller on incoming calls.

Outbound without P-Early Media support

During the call phase audio is negotiated and transmitted on outgoing calls.

The SIP header P-Early Media is not sent or evaluated.

Outbound with P-Early Media support (standard)

Audio is negotiated and transmitted on outgoing calls.

The SIP header P-Early Media is sent and evaluated.

If no P-Early Media header is received, EarlyMedia is used according to RFC 3261.

Outbound with P-Early Media and incoming only if P-Early Media is requested

The SIP header P-Early Media is sent and evaluated on outgoing calls.

On outgoing calls the SIP header P-Early Media is only sent and evaluated if it was received. Audio is transmitted.

This option only makes sense for sub-system operation and special accounts.

Outgoing and incoming

On outgoing and incoming calls, there is no limitation on negotiating EarlyMedia, the appropriate headers are considered.

Further help under External Internet Telephony

Use URI parameter (user)

Transfers the "User=Phone" parameter to the SIP header.

- Note: You may need to switch off this function if there are problems establishing the connection with a particular provider.
- Further help under External Internet Telephony

Use Mediasec header in acc. with IETF Draft

Transfers the SIP header in accordance with IETF draft "draft-dawes-sipcore-mediasec-parameter".

- Note: Some providers require this optional SIP header for establishing and running encrypted VoIP connections.
- Further help under External Internet Telephony

The provider has emergency call ability

It is possible to make an emergency call using this provider and the account assigned to it.

- Note: When switched off, this provider cannot be used for emergency calls. When you attempt to dial an emergency number, you will hear the announcement, "This telephone does not allow emergency calls. Please use an alternative."
- Further help under External Internet Telephony

Deregister, if there are NAT changes

The provider performs the deregistration and then the reregistration.

Some providers require this to take place every time a new public IP address has been assigned.

Further help under <u>External Internet Telephony</u>

DNS query outlasts SIP session (RFC 3263)

The SIP stack saves an entry for each information unit (PTR, SRV and AAAA) with that the preceded destination was reached. As long as the DNS server delivers this entry, it will be prioritised.

Further help under <u>External Internet Telephony</u>

Static public IP address

Some providers provide static public IP addresses and require them for call connection in the VIA header, the Contact header and in the SDP for the audio stream. Enter the provided IP address in the adjacent field and enable the option for the transmission (see previous options).

Further help under External Internet Telephony

Support T.38 for providers

T.38 enables smooth, extensive fax transmission to take place via set internal fax machines.

Note: If problems occur when T.38 is switched on, you can change the setting used to negotiate the fax protocol (T.38 negotiation).

Settings

Further help under External Internet Telephony

Number presentation (outgoing)

Provider

Select the provider to be displayed/configured.



Further help under External Internet Telephony

Format of called phone number

Format in which the provider requires the phone number of the destination, to enable the VoIP provider to transfer the call. The PBX automatically converts the phone numbers dialled into this format for calls made via the provider.

Country code with leading zeroes (e.g. 00495306...)

The destination phone number is transmitted from the PBX to the provider with the entire country code.

Country code with leading + sign (e.g. +495306...)

The destination phone number is transmitted from the PBX to the provider with the entire country code in mobile telephony format.

Country code without leading zeroes (e.g. 495306...)

The destination phone number is transmitted from the PBX to the provider with a shortened country code, without the international prefix (digit).

Without country code (e.g. 05306...)

The destination phone number is transmitted from the PBX to the provider without its own country code but with the local area code.

Local area code without leading zero (e.g. 5306...)

The destination phone number is transmitted to the provider with local area code without leading zero.

send as dialled

The PBX transmits the destination phone number to the provider, as dialled

- Note: If prefixes are necessary for converting, the prefixes set for the account are used.
- Further help under <u>Number presentation</u>

Format of own phone number

Format in which the provider requires the phone number. The PBX checks, whether and which one of the specified telephone numbers is to be transferred and converts this phone number automatically into the selected format.

Country code with leading zeroes (e.g. 0049...)

The phone number to be transmitted is transmitted with the entire country code to the provider.

Country code with leading + sign (e.g. +49...)

The phone number to be transmitted is transmitted to the provider with the entire country code in mobile telephony format.

Country code without leading zeroes (e.g. 495306...)

The phone number to be transmitted to the provider with a shortened country code (without international prefix (number).

Without country code (e.g. 05306...)

The phone number to be transmitted is transmitted to the provider without country and local area code.

Only MSN or main number and DDI

The phone number, that is to be transferred, is sent to the provider without prefixes (e.g. 920082 or 92008279).

Only DDI (only possible for PTP connection)

The phone number, that is to be transferred, is sent to the provider without prefixes and PBX base number (e.g. 8279).

Local area code without leading zero (e.g. 5306...)

The phone number to be transmitted is transmitted to the provider with local area code without leading zero.

- Note: If prefixes are necessary for converting, the prefixes set for the account are used.
- Further help under <u>Number presentation</u>

Format of CLIP no screening phone number

Format in which the provider requires the phone number. The PBX checks, whether CLIP co screening is enabled and which one of the specified telephone numbers is to be transferred and converts this phone number automatically into the selected format.

Country code with leading zeroes (e.g. 0049...)

The phone number to be transmitted is transmitted with the entire country code to the provider.

Country code with leading + sign (e.g. +49...)

The phone number to be transmitted is transmitted to the provider with the entire country code in mobile telephony format.

Country code without leading zeroes (e.g. 495306...)

Number presentation (outgoing)

The phone number to be transmitted to the provider with a shortened country code (without international prefix (number).

Without country code (e.g. 05306...)

The phone number to be transmitted is transmitted to the provider without country and local area code.

Local area code without leading zero (e.g. 5306...)

The phone number to be transmitted is transmitted to the provider with local area code without leading zero.

- Note: If prefixes are necessary for converting, the prefixes set for the account are used.
- Further help under Number presentation

Number presentation type

In the display text

In the user name

In accordance with RFC3325 with P-Asserted-Identity

In accordance with RFC3325 with P-Preferred-Identity

User-defined

Further help under <u>Number presentation</u>

Method of number presentation suppression

Area of the From header in which the provider expects to receive the number presentation suppression.

No display text

Number presentation (outgoing)

If number presentation suppression is switched on, an empty display text is sent in the From header.

Anonymous

The sent display text in the From header is "anonymous" if number presentation suppression is switched on.

User Anonymous

If number presentation suppression is switched on, both the display text and the user name section in the From header contain "anonymous".

- Note: Number presentation suppression in accordance with RFC 3325 using a privacy header is also used automatically.
- Further help under Number presentation

Evaluation of incoming SIP messages

Provider

Select the provider to be displayed/configured.

Further help under External Internet Telephony

Internationalise unknown phone numbers

If phone numbers that are not in the standard canonical format are received: (e.g. +495306... or 00495306...), they are converted into this format.

Further help under External Internet Telephony

Evaluate Connected Party Number (COPN)

Enabling this function prevents that a direct callback fails due to an incorrect depiction of the phone number when the call is answered.

Further help under External Internet Telephony

Set prefix for the phone number

Sets a prefix for the country code before the phone number (maximum 10 characters).

Further help under External Internet Telephony

Source for called number

Default

Selects an analysis type for phone numbers which will work for most providers.

Request Line

Forces the phone number to be evaluated using the Request Line.

To Header

Forces the phone number to be evaluated using the To Header.



Further help under External Internet Telephony

Analysis type

Default

Selects an analysis type for phone numbers which will work for most providers.

In accordance with RFC3325

Selects the phone number to be evaluated in accordance with RFC 3325 (RFC3325). This method of evaluating telephone numbers is used if the number presentation is guaranteed by security mechanisms.

User-defined

User-defined | Network-provided number

Network-proofed incoming phone number: The phone number to be evaluated is searched in the invite according to the configured order.

User-defined | User-provided number

Unproofed incoming phone number (corresponds to CLIP no screening): The phone number to be evaluated is searched in the invite according to the configured order. Optional: is not transferred by all providers.

User-defined | User-provided name

Caller's name in plain text. Optional: is not transferred by all providers.

Default (using the From Header as a Screened Number)

Selects an analysis type for phone numbers which will work for most providers. The user provided number (unscreened number, corresponds to CLIP no screening) in the From header is transferred to the network provided number (screened number). The phone numbers can be used, for example, for Call Through via VoIP.

- Caution: As soon as you have enabled this option, your telephone and PBX are open to attacks such as the transfer of manipulated phone numbers. The phone numbers of incoming calls listed in the call data list usually need to be converted before they can be used for callbacks.
- Further help under <u>External Internet Telephony</u>

Exchange line routing

Exchange line routing incoming Exchange line routing outgoing

Exchange line routing incoming

Block- and allowlist Call distribution for

Block- and allowlist

Overview Adding Configure

Overview

Blocklist

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for the user/group.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Allowlist

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for the user/group.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

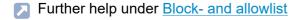
Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Adding

Name

Freely selectable name. This is used to identify this list within the PBX.



Type

Selection of a block- or allowlist for the incoming or outgoing exchange line routing.

Further help under Block- and allowlist

Rejecting anonymous calls

Anonymous incoming calls are rejected by the blocklist.

Further help under <u>Block- and allowlist</u>

Reject calls with special characters

The following characters are valid for phone numbers: 0-9, space and the following special characters: $(\) = +$

Phone numbers containing other characters will be rejected.

Further help under Block- and allowlist

Ext. Ext.number/range / Name

External phone numbers or phone number ranges and the corresponding name applicable to this list.

Adding

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

+ New

Creates a new entry for a block- or allowlist.

Further help under Block- and allowlist

Delete

Deletes the entries marked in the list.

Configure

Telephone number / range

External phone number or external phone number range applicable to this list.

Further help under Block- and allowlist

Name

Freely selectable name for the phone number or the phone number range.

Further help under Block- and allowlist

X Close

The dialogue for the configuration is closed, already created data is discarded.

+ New

Creates a new entry.

Save and close

The dialogue for the configuration is closed, already created data is saved.

iii Delete

Deletes the entries marked in the list.

Call distribution for

Overview Configure

Connection type | Account

Select the account to be displayed/configured.

Further help under Exchange line routing

Fallback on invalid telephone number

Calls directed to a non-existing number can be handled as follows:

Rejecting a call

The caller hears the busy signal, the call does not come internally.

Internal number

The caller is transferred to the selected internal number.

Further help under **Exchange line routing**

Transmitted SIP status for rejected calls (fallback and call distribution)

Sets the SIP status to be transmitted for rejected calls.

Further help under **Exchange line routing**

DDI/MSN

Number block/phone numbers that was configured for the selected account.

PBX connection: The following entries are created after the first setup:

• DDI**0** (e.g. for setting up the company headquarters)

- · Number block with the remaining available DDIs.
- Additional numbers
- + **New**: Creates a new entry for splitting an available number block.
- Further help under Exchange line routing

Routing rules

The methods set for the number block/phone number according to which incoming calls are distributed or diverted.

PBX connection: The linear call distribution is set during the initial setup.

Further help under Exchange line routing

Internal number | name

Internal number(s) and the associated name to which incoming calls are forwarded

Further help under Exchange line routing

Configure

Opens the configuration of the routing rule.

Details about settings under <u>Configure</u>

+ New

Not available for PTMP connections.

Creates a new entry for splitting an available number block.

Further help under Exchange line routing

Delete

Deletes the entries marked in the list.

Configure

Rules DDI / Range

Select the phone number block/phone number whose routing rules are to be configured.



Further help under Exchange line routing

Call distribution

Rejecting a call

The caller hears the busy signal, the call does not come internally.

Pass a call

The caller is passed to the internal destination that is entered in the following field.

Call Through

Not available for number blocks.

Defines the phone number as Call Through number that is used to call the PBX for Call Through or FMC.

Remote configuration

Not available for number blocks

Defines the phone number as remote access number that is used to call the PBX for remote programming.

- Further help under Exchange line routing
- Further help under Call Through

Configure

Further help under External access

Internal number | name

Only for selection Pass a call.

The internal number to which the caller is routed.

Further help under **Exchange line routing**

First internal destination

Only available for number blocks, if Pass a call.

Defines the beginning of the number range to which callers are forwarded linearly.

Further help under Exchange line routing

Fallback

Only available for number blocks, if Pass a call.

Calls directed to a non-existing number via the linear call distribution can be handled as follows:

Rejecting a call

The caller hears the busy signal, the call does not come internally.

Internal number

The caller is transferred to the selected internal number.

Further help under Exchange line routing

Call forwarding | delegate to an internal call destination

Not available for number blocks.

Requirement is an internal user as entered internal destination.

A for the entered user configured call forwarding is checked. If the call forwarding for the user has an external destination, the call forwarding is then already done in the exchange line for external calls.

Benefit: The telephone number of the external caller is transferred to the call forwarding destination.

Further help under Exchange line routing

Call forwarding | immediately (CFU)

Not available for number blocks.

If no reply, the caller is forwarded to the entered external destination number.

Further help under <u>Exchange line routing</u>

Call forwarding | if busy (CFB)

Not available for number blocks.

If busy, the caller is forwarded to the entered external destination number.

Further help under Exchange line routing

Call forwarding | no reply (CFNR)

Not available for number blocks.

Configure

If no reply, the caller is forwarded to the entered external destination number.



Further help under Exchange line routing

Exchange line routing outgoing

Overview Configure

Name

Freely selectable name. This is used to identify this rule set for outgoing exchange line routing within the PBX.

- Note: If necessary, important informations appears next to the name field. Such as e.g. in the event that no exchanges line capable of emergency calls have been set up.
- Further help under <u>Exchange line routing</u>

Configure

Opens the configuration for this rule set.

Details about settings under Configure

+ New

Creates a new rule set for outgoing exchange line routing.

Further help under Exchange line routing

m Delete

Deletes the entries marked in the list.

Configure

Configure

Block- and allowlist Routing

Block- and allowlist

Overview Adding Configure

Blocklist

The user/group cannot use the phone numbers entered in the selected blocklist for outgoing calls.

None

No blocklist is used for the user/group.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Allowlist

The phone numbers entered in the selected allowlist are freely available to the user/group for outgoing calls.

None

No allowlist is used for the user/group.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Adding

Name

Freely selectable name. This is used to identify this list within the PBX.

Further help under Block- and allowlist

Type

Selection of a block- or allowlist for the incoming or outgoing exchange line routing.

Further help under Block- and allowlist

Ext. Ext.number/range / Name

External phone numbers or phone number ranges and the corresponding name applicable to this list.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

+ New

Creates a new entry for a block- or allowlist.

Further help under Block- and allowlist



Deletes the entries marked in the list.

Configure

Telephone number / range

External phone number or external phone number range applicable to this list.

Further help under Block- and allowlist

Name

Freely selectable name for the phone number or the phone number range.

Further help under Block- and allowlist

X Close

The dialogue for the configuration is closed, already created data is discarded.

+ New

Creates a new entry.

Save and close

The dialogue for the configuration is closed, already created data is saved.

iii Delete

Deletes the entries marked in the list.

Routing

Designation

This is used to identify the routing rules within the PBX.

Emergency calls

Part of each rule set. The emergency numbers set up on the PBX are used for emergency calls. These calls are handled in a special way.

International

Part of each rule set. Includes all international phone numbers.

National

Part of each rule set. Includes all national phone numbers.

City

Part of each rule set. Includes all phone numbers without prefix.

Custom routing

Freely selectable name.

- Further help under **Emergency calls**
- Further help under Exchange line routing

Telephone number / range

Only available for user-defined routing.

Defines for which phone number/number range the routing rule applies.

Further help under <u>Exchange line routing</u>

1st to 3rd dialling

Defines the account to be used for dialling depending on the dialled number.

all accounts

The dial attempt is made over one of the configured VoIP accounts. If there is an account on site, this is used by the PBX first.

[Name of the account]

The dial attempt is made over the VoIP account permanently assigned. In case that the preferred account is busy, an alternative for a second or third dialling can be defined.

send busy

The dial attempt is ended and a busy signal is issued. If this has already been selected for the first dialling, there is no exchange line authorisation for the according phone number/number range/number category.



Further help under Exchange line routing

Number presentation

Defines the number presentation depending on the dialled phone number. The setting can be overridden by the individual setting **Transmit the own number** of the user/group.

free input

The entered phone number (e.g. a service number) is tranferred. Requirement for this application is that CLIP no screening is supported and enabled on the account

[Phone number | name]

Routing

The selected phone number is transmitted. Selecting a number block requires the entry of an extension (DDI).

Further help under **Exchange line routing**

+ New

Creates a new routing rule for the exchange line routing rule set.

Further help under Exchange line routing

m Delete

Deletes the entries marked in the list.

User

User

Phone numbers
Profiles
Functional Overview

Phone numbers

Overview Configure

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by another internal user.

- Note: The internal number entered here needs to be entered in the telephone as the (first) MSN.
- Further help under <u>Internal numbers</u>
- Further help under <u>User</u>

Name

Freely selectable name. This name is used to identify the user within the PBX and from connected phones. The name can also be transferred to the VoIP end devices via the LDAP server.

Further help under <u>User</u>

Configure

Opens the configuration of the user.

Details about settings under Configure

Filter

The list is filtered by the entered or selected value.

+ New

Adds a new user. You must then make the settings for this new user.

Further help under <u>User</u>

m Delete

Deletes the entries marked in the list.

Configure

Basic Settings
Exchange line settings
Reachability
Authorisations
Voice mail and fax boxes
Settings

Basic Settings

User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

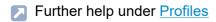
In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



PIN

Automatically generated, freely changeable PIN. Is used, among other things, to access the voice mail and fax box and for the functions FMC and Roaming User.

Note: To change the existing PIN, click on

Edit PIN and password. After you have entered your PIN or password, click on

Hide PIN and password.

Further help under <u>Access authorisations</u>

Password

Automatically generated, freely changeable password. Currently not used.

- Note: To change the current password, click on

 Edit PIN and password. After you have entered your PIN or password, click on

 Hide PIN and password.
- Further help under <u>Access authorisations</u>

Additional passwords | - Configure

Opens the configuration of additional passwords.

Requirement is the enabled function Users may generate own passwords for accessing the web interface / API (page User > Functional Overview)

- Note: If the administrator has made the settings mentioned above, the user can configure additional passwords for logging in to the web interface / API. The administrator may still create additional passwords for users.
- Note: The last password to use the web interface/API cannot be deleted.
- Note: After the user has assigned another password, he cannot go back to the original password. Only the administrator has this authorisation.
- Further help under <u>Access authorisations</u>

Description

Freely selectable name for this additional password.

Further help under <u>Access authorisations</u>

User

Phone number and name of the user the additional password was created for.

Further help under Access authorisations

API Access

The user has access to the web interface / API.

Further help under Access authorisations

× Close

The dialogue for the configuration is closed, already created data is discarded.

+ New

Creates a new entry.

Further help under Access authorisations

☐ Save and close

The dialogue for the configuration is closed, already created data is saved.



Deletes the entries marked in the list.

Logged in to the web interface

Displays existing logins of the user to the administrator.

Deregister

Logs out the user from the web interface.

Further help under <u>Access authorisations</u>

Exchange line settings

User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Prevention of emergency calls

Emergency calls cannot be made. The user hears a corresponding announcement.

Further help under Emergency calls

Transmit the own number

Assigns an account, and therefore also a location, to the user. You can select the accounts that have been assigned to the user in the call distribution.

Defines the phone number of the assigned account as the phone number that will handle outgoing phone numbers.



Further help under <u>Number presentation</u>

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.



Further help under Exchange line routing

Reachability

User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.

Further help under Profiles

Call waiting

During a call, incoming calls are signalled by a call waiting tone.

Further help under <u>Call waiting</u>

Do-not-Disturb

Callers are rejected.

Reachability

Further help under <u>Do-not-Disturb</u>

Blocklist | Incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for the user.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under Configure

Allowlist | Incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for the user.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under <u>Block- and allowlist</u>

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Follow-me (internal/external) | Destination number:

Internal and external calls to the user are forwarded to the entered destination number.

Further help under Follow-me

Multi-path call forwarding

Multi-path call forwarding behaviour | Destination number

- Internal and external calls directed to the user are also signalled at the entered destination number.
- · Call forwarding on no reply is executed
- Calls directed to a group are not considered
- Further help under <u>Multi-path call forwarding</u>

Call forwarding | Settings | only external calls

External calls to the user are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding | Settings | CF on group call

Calls directed to a group in which the user is logged in as incoming will be forwarded to the CF destination of the user.

Further help under Call forwarding

Call forwarding | CF immediately | Destination number:

Internal and external calls to the user are forwarded immediately to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on busy | Destination number:

Internal and external calls to the user are forwarded on busy to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the user are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.



Further help under Call forwarding

Authorisations

User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Pick-up

No

Calls of the user may not be taken by pick-up.

Global

The user may take calls to all other users by pick-up.

Group(s)/voice mailbox(es)

Authorisations

This user may take calls of those users that are members of groups the user is also member of. It does not make any difference if the user is currently logged-in or -out. The user is authorised to take over calls from those voice mailboxes of which he is the owner.

If a voice mailbox has a group as its owner or user, each user who is a member of the group can use this authorisation to take over calls.

Further help under Pick-Up and Call Pick-Up

Configuration switchover

The user can switch configurations manually.

Further help under Configuration switchover

Loudspeaker Audio Out

No

The user may not perform speaker announcements.

[Name]

The user can perform speaker announcements on the selected announcement output.

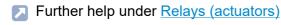
All announcement outputs

The user can perform speaker announcements on all announcement outputs.

Further help under Loudspeaker Audio Out

Relay switching

The user can switch relays manually.



Controlling of the PBX via phone

The user may change settings of the PBX (e.g. Do-not-Disturb) using a menu/function key or a character string.

Further help under Access authorisations

Opening door

Nο

The user may not open a door.

[Name]

The user can phone with the selected door and open the door by pressing a key.

All doors

The user can phone with all doors and open the doors by pressing a key.

Further help under **Door functions**

Features to be configured by the network provider (CF, key pad)

The user can

forward the own external number via phone.

Authorisations

- · send keypad information to the exchange line.
- Further help under Call forwarding

Configuring CF (sub) to ext. number

The user can forward the associated external number to an external destination.

Further help under <u>Call forwarding</u>

Deleting the List of Single Call Records

The user can delete the phone number of the communication partner from the last call made to ensure that this number is not viewed by unauthorised persons.

Further help under Call data

CF (groups) / Follow-me

The user can

- · switching group properties of groups which the user is a member of.
- switch on Follow-me.
- Further help under Follow-me

Call on hold

The user can park a caller in the parking zone or retrieve a waiting caller from the parking zone.

Authorisations

Further help under Call on hold

Roaming User

The user can log onto any telephone that is available to them and release the telephones used by them again.



Further help under Roaming User

Voice mail and fax boxes

User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.

Further help under <u>Profiles</u>

Fax box | Telephone number of the box

A created fax box for the user is required in order to have the possibility of selecting a telephone number.

- Note: If it is not clear whether it is a fax or not, the call will be signalled to the user or it will be accepted by the voice mailbox.
- Note: A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.

Further help under Voice mail and fax boxes

Voice mailbox | Telephone number of the box

A created voice mailbox for the user is required in order to have the possibility of selecting a telephone number.

Both the user and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the user.

- Note: If call forwarding immediately (user/group) is enabled, the call will not be made to the voice mailbox.
- Further help under Voice mail and fax boxes

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the user is busy and if no call forwarding on busy has been enabled for the user.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the user does not reply or if the internal number of the user is busy.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here.



User

Select the user to be displayed/configured.

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Preliminary pick-up query and monitoring

Requirements:

VoIP end device which can evaluate the available information by BLF

All information about the calls of the user is available for evaluation by BLF.

A preliminary pick-up query allows a user, even before a pick-up, to see which caller he would be picking up.

- Note: When switched off, no phone numbers of the user's calls will be transmitted by BLF.
- Further help under Preliminary pick-up query and monitoring

Call pick-up

Nο

Existing calls of this user may not be taken over.

Global

Existing calls of this user may be taken over by every other phone.

Group(s)

Existing calls of this user may be taken over by members of groups the user is also member of. It does not make any difference if the user is currently logged-in or -out.

Further help under Pick-Up and Call Pick-Up

Central caller list

The PBX saves the user's data in a central caller list and makes this available on the user's telephone.

Further help under <u>Central caller list</u>

Intercom permission

Any of the user's telephones that can evaluate the SIP header for Auto Answer automatically accept an InterCom that is sent to that user.



Further help under InterCom

Profiles

Profiles

Profile management
Profile Assignment
Configure

Profile management

Name

The available profiles (default profiles, user-defined profiles) are displayed.

Further help under Profiles

View

The selected settings can be viewed (default profiles).

Further help under Profiles

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

Details about settings under <u>Configure</u>

+New

Creation of a new profile.

Further help under Profiles

Profile Assignment

Assignment

The selected profile is assigned to the user, the group or the box.



All

The same profile is assigned to all users, groups or boxes.

Further help under Profiles

Configure

Exchange line settings
Reachability
Authorisations
Voice mailbox
Settings

Exchange line settings

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

Prevention of emergency calls

Emergency calls cannot be made. The user hears a corresponding announcement.

Further help under <u>Emergency calls</u>

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Exchange line settings



Further help under **Exchange line routing**

Reachability

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- **Note:** When switched off, the setting made for the user is valid.
- Further help under Profiles

Blocklist | Incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for the user.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Reachability

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Allowlist | Incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for the user.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under Configure

Multi-path call forwarding

Multi-path call forwarding behaviour | Destination number

- Internal and external calls directed to the user are also signalled at the entered destination number.
- · Call forwarding on no reply is executed
- Calls directed to a group are not considered
- Further help under Multi-path call forwarding

Call forwarding | Settings | only external calls

External calls to the user are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding | Settings | CF on group call

Calls directed to a group in which the user is logged in as incoming will be forwarded to the CF destination of the user

Further help under Call forwarding

Call forwarding | CF immediately | Destination number:

Internal and external calls to the user are forwarded immediately to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on busy | Destination number:

Internal and external calls to the user are forwarded on busy to the entered destination number.

Further help under <u>Call forwarding</u>

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the user are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Further help under Call forwarding

Authorisations

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

Relay switching

The user can switch relays manually.

Further help under Relays (actuators)

Controlling of the PBX via phone

The user may change settings of the PBX (e.g. Do-not-Disturb) using a menu/function key or a character string.

Further help under <u>Access authorisations</u>

Opening door

No

The user may not open a door.

[Name]

The user can phone with the selected door and open the door by pressing a key.

All doors

The user can phone with all doors and open the doors by pressing a key.

Further help under Door functions

Features to be configured by the network provider (CF, key pad)

The user can

- forward the own external number via phone.
- send keypad information to the exchange line.
- Further help under Call forwarding

Configuring CF (sub) to ext. number

The user can forward the associated external number to an external destination.

Further help under <u>Call forwarding</u>

CF (groups) / Follow-me

The user can

- switching group properties of groups which the user is a member of.
- · switch on Follow-me.
- Further help under Follow-me

Voice mailbox

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the user is busy and if no call forwarding on busy has been enabled for the user.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the user does not reply or if the internal number of the user is busy.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here.

Further help under Voice mail and fax boxes

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under **Profiles**

Profile-controlled settings

The profile setting overrides the user setting.

- **Note:** When switched off, the setting made for the user is valid.
- Further help under Profiles

Preliminary pick-up query and monitoring

Requirements:

VoIP end device which can evaluate the available information by BLF

All information about the calls of the user is available for evaluation by BLF.

A preliminary pick-up query allows a user, even before a pick-up, to see which caller he would be picking up.

- **Note:** When switched off, no phone numbers of the user's calls will be transmitted by BLF.
- Further help under Preliminary pick-up query and monitoring

Functional Overview

Global settings Functional Overview

Global settings

Max. duration of switched calls: external with external

Specifies how many minutes pause there is until the PBX ends the transferred call.

Further help under Query and Transfer

Ringer frequency

Specifies the ringer voltage frequency the PBX uses to signalise calls to internal analogue users.

Users may generate own passwords for accessing the web server / API

The user may create additional passwords or delete existing passwords.

- **Note:** When switched off, only the administrator may create additional passwords.
- Further help under Access authorisations

Functional Overview

Basic Settings
Exchange line settings
Reachability
Authorisations
Voice mail and fax boxes
Settings

Basic Settings

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

PIN

Automatically generated, freely changeable PIN. Is used, among other things, to access the voice mail and fax box and for the functions FMC and Roaming User.

- Note: To change the existing PIN, click on

 Edit PIN and password. After you have entered your PIN or password, click on

 Hide PIN and password.
- Further help under Access authorisations

Password

Automatically generated, freely changeable password. Currently not used.

- Note: To change the current password, click on

 Edit PIN and password. After you have entered your PIN or password, click on

 Hide PIN and password.
- Further help under <u>Access authorisations</u>

Exchange line settings

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- **Note:** When switched off, the setting made for the user is valid.
- Further help under Profiles

Prevention of emergency calls

Emergency calls cannot be made. The user hears a corresponding announcement.

Further help under <u>Emergency calls</u>

Transmit the own number

Assigns an account, and therefore also a location, to the user. You can select the accounts that have been assigned to the user in the call distribution.

Exchange line settings

Defines the phone number of the assigned account as the phone number that will handle outgoing phone numbers.



Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Further help under Exchange line routing

Reachability

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under <u>Profiles</u>

Call waiting

During a call, incoming calls are signalled by a call waiting tone.

Further help under <u>Call waiting</u>

Do-not-Disturb

Callers are rejected.

Further help under **Do-not-Disturb**

Blocklist | Incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for the user.

Existing blocklists

Selection of an already created blocklist.

Further help under Block- and allowlist

Allowlist | Incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for the user.

Existing allowlists

Selection of an already created allowlist.

Further help under Block- and allowlist

Follow-me (internal/external) | Destination number:

Internal and external calls to the user are forwarded to the entered destination number.

Further help under Follow-me

Multi-path call forwarding

Multi-path call forwarding behaviour | Destination number

- Internal and external calls directed to the user are also signalled at the entered destination number.
- Call forwarding on no reply is executed
- · Calls directed to a group are not considered
- Further help under Multi-path call forwarding

Call forwarding | Settings | only external calls

External calls to the user are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding | Settings | CF on group call

Calls directed to a group in which the user is logged in as incoming will be forwarded to the CF destination of the user.

Further help under Call forwarding

Call forwarding | CF immediately | Destination number:

Internal and external calls to the user are forwarded immediately to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on busy | Destination number:

Internal and external calls to the user are forwarded on busy to the entered destination number.

Further help under <u>Call forwarding</u>

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the user are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Further help under Call forwarding

Authorisations

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

Pick-up

No

Calls of the user may not be taken by pick-up.

Global

The user may take calls to all other users by pick-up.

Group(s)/voice mailbox(es)

This user may take calls of those users that are members of groups the user is also member of. It does not make any difference if the user is

Authorisations

currently logged-in or -out. The user is authorised to take over calls from those voice mailboxes of which he is the owner.

If a voice mailbox has a group as its owner or user, each user who is a member of the group can use this authorisation to take over calls.

Further help under Pick-Up and Call Pick-Up

Configuration switchover

The user can switch configurations manually.

Further help under Configuration switchover

Loudspeaker Audio Out

No

The user may not perform speaker announcements.

[Name]

The user can perform speaker announcements on the selected announcement output.

All announcement outputs

The user can perform speaker announcements on all announcement outputs.

Further help under <u>Loudspeaker Audio Out</u>

Relay switching

The user can switch relays manually.

Further help under Relays (actuators)

Controlling of the PBX via phone

The user may change settings of the PBX (e.g. Do-not-Disturb) using a menu/function key or a character string.

Further help under Access authorisations

Opening door

No

The user may not open a door.

[Name]

The user can phone with the selected door and open the door by pressing a key.

All doors

The user can phone with all doors and open the doors by pressing a key.

Further help under Door functions

Features to be configured by the network provider (CF, key pad)

The user can

- forward the own external number via phone.
- send keypad information to the exchange line.
- Further help under <u>Call forwarding</u>

Configuring CF (sub) to ext. number

The user can forward the associated external number to an external destination.

Further help under Call forwarding

Deleting the List of Single Call Records

The user can delete the phone number of the communication partner from the last call made to ensure that this number is not viewed by unauthorised persons.

Further help under Call data

CF (groups) / Follow-me

The user can

- switching group properties of groups which the user is a member of.
- switch on Follow-me
- Further help under Follow-me

Call on hold

The user can park a caller in the parking zone or retrieve a waiting caller from the parking zone.

Further help under <u>Call on hold</u>

Roaming User

The user can log onto any telephone that is available to them and release the telephones used by them again.



Further help under Roaming User

Voice mail and fax boxes

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under Profiles

Fax box | Telephone number of the box

A created fax box for the user is required in order to have the possibility of selecting a telephone number.

- Note: If it is not clear whether it is a fax or not, the call will be signalled to the user or it will be accepted by the voice mailbox.
- Note: A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.
- Further help under Voice mail and fax boxes

Voice mailbox | Telephone number of the box

A created voice mailbox for the user is required in order to have the possibility of selecting a telephone number.

Both the user and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the user.

- Note: If call forwarding immediately (user/group) is enabled, the call will not be made to the voice mailbox.
- Further help under Voice mail and fax boxes

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on busy

The voice mailbox will be called if the internal number of the user is busy and if no call forwarding on busy has been enabled for the user.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on no reply + on busy

The voice mailbox will be called if the user does not reply or if the internal number of the user is busy.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here.



Settings

Property of the user

Opens the configuration of the single function for all users.

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the user setting.

- Note: When switched off, the setting made for the user is valid.
- Further help under <u>Profiles</u>

Preliminary pick-up query and monitoring

Requirements:

· VoIP end device which can evaluate the available information by BLF

All information about the calls of the user is available for evaluation by BLF.

A preliminary pick-up query allows a user, even before a pick-up, to see which caller he would be picking up.

Note: When switched off, no phone numbers of the user's calls will be transmitted by BLF.

Settings

Further help under <u>Preliminary pick-up query and monitoring</u>

Call pick-up

No

Existing calls of this user may not be taken over.

Global

Existing calls of this user may be taken over by every other phone.

Group(s)

Existing calls of this user may be taken over by members of groups the user is also member of. It does not make any difference if the user is currently logged-in or -out.

Further help under Pick-Up and Call Pick-Up

Central caller list

The PBX saves the user's data in a central caller list and makes this available on the user's telephone.

Further help under Central caller list

Intercom permission

Any of the user's telephones that can evaluate the SIP header for Auto Answer automatically accept an InterCom that is sent to that user.

Further help under InterCom

Groups

Groups

Phone numbers
Profiles
Functional Overview

Phone numbers

Overview Configure

Overview

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by an internal user.

- Further help under <u>Internal numbers</u>
- Further help under Groups

Name

Freely selectable name. This name is used to identify the group within the PBX and the connected phones. The name can also be transferred to the VoIP end devices via the LDAP server

Further help under Groups

Group members

Displays all users belonging to the group and groups.

Further help under Groups

Incoming

Logged in

Logs the telephone in to the group.

Logged out

Logs the telephone out of the group.

Permanently logged in

Permanently logs the phone of a subgroup into the main group.



Outgoing

Logged in

Logs the telephone in to the group.

Logged out

Logs the telephone out of the group.

Permanently logged in

Permanently logs the phone of a subgroup into the main group.

Further help under Groups

Configure

Opens the configuration of the group.

- Details about settings under Configure
- Further help under Groups

Filter

The list is filtered by the entered or selected value.

+ New

Adds a new group. You must then make the settings for this new group.

Overview

Further help under Groups

m Delete

Deletes the entries marked in the list.

Configure

Basic Settings
Exchange line settings
Reachability
Voice mail and fax boxes
Settings

Basic Settings

Group

Select the group you want to display/configure.



Further help under Groups

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Call Strategy

ΔII

All users logged in to the group will ring simultaneously. Selected users will start to ring after the configured delay time (after:). If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

All - Split Group

The call is handled almost similar to the call strategy **AII**. The only difference is the way multiple calls to the group are handled: If the group receives further calls during an active call, the group will be split into several smaller groups ringing for a specific active call. But there will never ring more calls in the group than the number of free available group members. Additional calls will then be rejected with a busy. If a call is accepted, the other phones ringing so far for this call are assigned to the other pending calls and will immediately start to ring again. If a group member finishes a call, this user will immediately ring again for the pending calls. The timer for the configured delay time (**after:**) starts after receiving the first call to the group and will not restart for additional incoming calls.

Linear - fixed - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - balancing - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - fixed - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**.

Note: System phones control the call waiting itself, so the call strategy can vary.

Linear - balancing - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**: The first loggedin group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call.

Adding - fixed - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the

remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - balancing - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

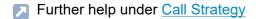
Adding - fixed - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already

ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If finally all logged-in members of a group are called, the busy users will hear knocking additionally (only from external).

Adding - balancing - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If finally all loggedin members of a group are called, the busy users will hear knocking additionally (only from external).



Phone number | Name

Group member. (Entry with + **New**)

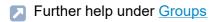
Further help under Groups

Priority

Requirements:

Configured call strategy: NotAll or All - Split Group

Specifies the order of calls within the group for the call strategy (change with $\sim | \searrow$).



Incoming

Logged in

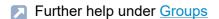
Logs the telephone in to the group.

Logged out

Logs the telephone out of the group.

Permanently logged in

Permanently logs the phone of a subgroup into the main group.



Outgoing

Logged in

Logs the telephone in to the group.

Logged out

Logs the telephone out of the group.

Permanently logged in

Basic Settings

Permanently logs the phone of a subgroup into the main group.

Further help under Groups

Ringing delay

Requirements:

Configured call strategy: All or All - Split Group

The user rings after the configured delay time (after:).

Further help under Groups

+ New

Selects a new group member from the configured users and groups and adds it to the list.

Further help under **Groups**

m Delete

Deletes the entries marked in the list.

Exchange line settings

Group

Select the group you want to display/configure.



Further help under Groups

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.



Further help under **Exchange line routing**

Own number

Assigns an account, and therefore also a location, to the group. You can select the accounts that have been assigned to the group in the call distribution.

Further help under Number presentation

Transmit the own number

Defines the phone number of the assigned account as the phone number that will handle outgoing phone numbers.

Further help under <u>Number presentation</u>

Reachability

Group

Select the group you want to display/configure.



Further help under Groups

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Reserved call channels

Number of reserved external VoIP channels for the group. The reservation can only be made on one account and the associated provider. These call channels are only available for users who are logged in this group as outgoing.



Further help under Reserved call channels

Busy-on-Busy

Callers are put through depending on the following factors:

- Number of available group members: The number is determined for each call. Incoming logged-in group members with at least one registered telephone are taken into account.
- Number of active connections: The number is made up of active connections of the group members and (if announcement before answering is enabled) callers who were automatically answered and connected to the waiting loop.

deactivated

Callers get busy as soon as the number of active connections equals the number of available group members.

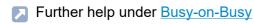
Exception: If announcement before answering is enabled, the reachability of the group and thus the calls connected to the waiting loop is unlimited.

dynamically

Callers get busy as soon as the number of active connections equals the number of available group members.

busy from x connections on

Callers get busy as soon as the number of connections set here is reached.



Announcement before answering

Off

When calling a group, no announcement is played. Available group members are immediately called. If all group members are busy, the caller hears the busy tone.

On busy

When calling a group, an announcement is played only if busy. Available group members are immediately called. If all group members are busy, an announcement is played for greeting. After the announcement, the caller is put on hold and hears Music on Hold.

Always

When calling a group the caller always hears the announcement for greeting. If group members are available, they are called. If all the group members are busy, the caller is put on hold after the announcement.



Announcement before answering | Call signalling

Immediately

The call is signalled during the announcement.

After Announcement before Answering

The call is signalled after the announcement when the caller hears Music on Hold.

Further help under Announcement before answering

Announcement before answering | Selection of announcement

Announcement which is played for the caller as greeting.

Further help under <u>Announcement before answering</u>

Rework time

Time that a group member cannot be called after an incoming internal or external call (e.g. for call rework). During this time, the Do-not-Disturb function is enabled for the user.

- Note: The rework time can be cancelled at any time by switching off the Do-not-Disturb function (e.g. using a function key at the system phone).
- **Note:** The following characteristics apply for the devices COMfortel 3200/3500: Enabling this function on the telephone is not displayed in the web interface of the PBX. Enabling this function in the web interface of the PBX has no influence on the telephone (status is not displayed under the function keys)

Blocklist | incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for group members logged in as outgoing.

Like user

The blocklists selected for the users are used.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Reachability

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Allowlist | incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for group members logged in as outgoing.

Like user

The allowlists selected for the users are used.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Follow-me (internal/external) | Destination number:

Internal and external calls to the group are forwarded to the entered destination number.

Further help under Follow-me

Multi-path call forwarding | Destination number:

Internal and external calls to the group are signalled additionally to the entered destination number.

- Note: Multi-path Call Forwarding for a group is executed during the entire calling period independent of the call strategy for the group. It is not executed if the group itself is the destination of a forwarded call.
- Further help under Multi-path call forwarding

Call forwarding | Settings | only external calls

External calls to the group are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding |Settings | CF on group call

Calls directed to a group of which the group is a member will be forwarded to the CF destination of the group.

Further help under <u>Call forwarding</u>

Call forwarding | CF immediately | Destination number:

Internal and external calls to the group are forwarded immediately to the entered destination number.

Further help under <u>Call forwarding</u>

Call forwarding | CF on busy | Destination number:

Internal and external calls to the group are forwarded on busy to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the group are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Further help under Call forwarding

Call forwarding | on all logged out | Destination number:

Internal and external calls to the group are forwarded to the entered destination number, if all group members are logged out.

Further help under <u>Call forwarding</u>

Voice mail and fax boxes

Group

Select the group you want to display/configure.



Further help under Groups

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under **Profiles**

Fax box | Telephone number of the box

A created fax box for the group is required in order to have the possibility of selecting a telephone number.

Note: If it is not clear whether it is a fax or not, the call will be signalled to the group or it will be accepted by the voice mailbox.

- Note: If a fax call is accepted by the group, it can be forwarded to a fax box by means of transfer.
- Note: A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.
- Further help under Voice mail and fax boxes

Voice mailbox | Telephone number of the box

Both the group and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the group.

- Note: If call forwarding immediately (user/group) is enabled, the call will not be made to the voice mailbox.
- Further help under Voice mail and fax boxes

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on busy

The voice mailbox will be called if the internal number of the group is busy and if no call forwarding on busy has been enabled for the group.

on no reply + on busy

The voice mailbox will be called if the group does not reply or if the internal number of the group is busy.

Voice mail and fax boxes

Further help under Voice mail and fax boxes

On all logged out

The voice mailbox will be called when all group members are logged out.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here in sec.

Further help under Voice mail and fax boxes

Settings

Group

Select the group you want to display/configure.



Further help under Groups

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Number presentation (internally)

Group numbers

All group members logged in as outgoing will present the group number for internal outgoing calls. If the group is internally called, the user number is transferred back.

User phone number

Settings

All group members logged in as outgoing will present their own user number for internal outgoing calls.

Further help under Number presentation

Preliminary pick-up query and monitoring

Requirement is a VoIP end device which can evaluate the available information by BLF.

All information about the incoming calls of the group is available for evaluation by BLF.

- Note: When switched off, no phone numbers of the group's calls are transmitted by BLF.
- Further help under Preliminary pick-up query and monitoring

Pick-up of all calls to group members

A member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

- **Note:** When switched off, group members can only pick up direct calls to the phone number of the group.
- Further help under Pick-up of all calls to group members

Profiles

Profiles

Profile management
Profile Assignment
Configure

Profile management

Name

The available profiles (default profiles, user-defined profiles) are displayed.

Further help under Profiles

View

The selected settings can be viewed (default profiles).

Further help under Profiles

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

Details about settings under <u>Configure</u>

+New

Creation of a new profile.

Further help under Profiles

Profile Assignment

Assignment

The selected profile is assigned to the user, the group or the box.



All

The same profile is assigned to all users, groups or boxes.

Further help under Profiles

Configure

Basic Settings
Exchange line settings
Reachability
Voice mail and fax boxes
Settings

Basic Settings

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Call Strategy

ΑII

All users logged in to the group will ring simultaneously. Selected users will start to ring after the configured delay time (**after:**). If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

All - Split Group

The call is handled almost similar to the call strategy **AII**. The only difference is the way multiple calls to the group are handled: If the group receives further calls during an active call, the group will be split into several smaller groups ringing for a specific active call. But there will never ring more calls in the group than the number of free available group members. Additional calls will then be rejected with a busy. If a call is accepted, the other phones ringing so far for this call are assigned to the other pending calls and will immediately start to ring again. If a group member finishes a call, this user will immediately ring again for the pending calls. The timer for the configured delay time (**after:**) starts after receiving the first call to the group and will not restart for additional incoming calls.

Linear - fixed - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - balancing - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - fixed - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**.

Note: System phones control the call waiting itself, so the call strategy can vary.

Linear - balancing - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**: The first loggedin group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call.

Adding - fixed - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already

ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - balancing - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - fixed - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends

on the configured **Priority**. If finally all logged-in members of a group are called, the busy users will hear knocking additionally (only from external).

Adding - balancing - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If finally all loggedin members of a group are called, the busy users will hear knocking additionally (only from external).

Further help under Call Strategy

Exchange line settings

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- Note: When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Further help under Exchange line routing

Reachability

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- Note: When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Reserved call channels

Number of reserved external VoIP channels for the group. The reservation can only be made on one account and the associated provider. These call channels are only available for users who are logged in this group as outgoing.

Further help under Reserved call channels

Busy-on-Busy

Callers are put through depending on the following factors:

- Number of available group members: The number is determined for each call. Incoming logged-in group members with at least one registered telephone are taken into account.
- Number of active connections: The number is made up of active connections of the group members and (if announcement before answering is enabled) callers who were automatically answered and connected to the waiting loop.

deactivated

Callers get busy as soon as the number of active connections equals the number of available group members.

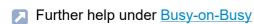
Exception: If announcement before answering is enabled, the reachability of the group and thus the calls connected to the waiting loop is unlimited.

dynamically

Callers get busy as soon as the number of active connections equals the number of available group members.

busy from x connections on

Callers get busy as soon as the number of connections set here is reached.



Announcement before answering

Off

When calling a group, no announcement is played. Available group members are immediately called. If all group members are busy, the caller hears the busy tone.

On busy

When calling a group, an announcement is played only if busy. Available group members are immediately called. If all group members are busy, an announcement is played for greeting. After the announcement, the caller is put on hold and hears Music on Hold.

Always

When calling a group the caller always hears the announcement for greeting. If group members are available, they are called. If all the group members are busy, the caller is put on hold after the announcement.



Announcement before answering | Call signalling

Immediately

The call is signalled during the announcement.

After Announcement before Answering

The call is signalled after the announcement when the caller hears Music on Hold.

Further help under Announcement before answering

Announcement before answering | Selection of announcement

Announcement which is played for the caller as greeting.

Reachability

Further help under Announcement before answering

Blocklist | incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for group members logged in as outgoing.

Like user

The blocklists selected for the users are used.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Allowlist | incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for group members logged in as outgoing.

Like user

The allowlists selected for the users are used.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under Configure

Multi-path call forwarding | Destination number:

Internal and external calls to the group are signalled additionally to the entered destination number.

- Note: Multi-path Call Forwarding for a group is executed during the entire calling period independent of the call strategy for the group. It is not executed if the group itself is the destination of a forwarded call.
- Further help under Multi-path call forwarding

Call forwarding | Settings | only external calls

External calls to the group are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding |Settings | CF on group call

Calls directed to a group of which the group is a member will be forwarded to the CF destination of the group.

Further help under Call forwarding

Call forwarding | CF immediately | Destination number:

Internal and external calls to the group are forwarded immediately to the entered destination number.

Further help under <u>Call forwarding</u>

Call forwarding | CF on busy | Destination number:

Internal and external calls to the group are forwarded on busy to the entered destination number.

Further help under <u>Call forwarding</u>

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the group are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Further help under Call forwarding

Call forwarding | on all logged out | Destination number:

Internal and external calls to the group are forwarded to the entered destination number, if all group members are logged out.



Further help under Call forwarding

Voice mail and fax boxes

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- Note: When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on busy

The voice mailbox will be called if the internal number of the group is busy and if no call forwarding on busy has been enabled for the group.

on no reply + on busy

The voice mailbox will be called if the group does not reply or if the internal number of the group is busy.

Further help under Voice mail and fax boxes

On all logged out

The voice mailbox will be called when all group members are logged out.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here in sec.

Further help under Voice mail and fax boxes

Settings

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Number presentation (internally)

Group numbers

All group members logged in as outgoing will present the group number for internal outgoing calls. If the group is internally called, the user number is transferred back.

User phone number

Settings

All group members logged in as outgoing will present their own user number for internal outgoing calls.

Further help under <u>Number presentation</u>

Preliminary pick-up query and monitoring

Requirement is a VoIP end device which can evaluate the available information by BLF.

All information about the incoming calls of the group is available for evaluation by BLF.

- Note: When switched off, no phone numbers of the group's calls are transmitted by BLF.
- Further help under Preliminary pick-up query and monitoring

Pick-up of all calls to group members

A member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

- **Note:** When switched off, group members can only pick up direct calls to the phone number of the group.
- Further help under Pick-up of all calls to group members

Functional Overview

Basic Settings
Exchange line settings
Reachability
Voice mail and fax boxes
Settings

Basic Settings

Property of the group

Opens the configuration of the single function for all groups.

Further help under Groups

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Call Strategy

ΑII

All users logged in to the group will ring simultaneously. Selected users will start to ring after the configured delay time (**after:**). If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

All - Split Group

The call is handled almost similar to the call strategy **AII**. The only difference is the way multiple calls to the group are handled: If the group receives further calls during an active call, the group will be split into several smaller groups ringing for a specific active call. But there will never ring more calls in the group than the number of free available group members. Additional calls will then be rejected with a busy. If a call is accepted, the other phones ringing so far for this call are assigned to the other pending calls and will immediately start to ring again. If a group member finishes a call, this user will immediately ring again for the pending calls. The timer for the configured delay time (**after:**) starts after receiving the first call to the group and will not restart for additional incoming calls.

Linear - fixed - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - balancing - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Linear - fixed - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**.

Note: System phones control the call waiting itself, so the call strategy can vary.

Linear - balancing - rotating - split group

Only the phone of one group member will ring. After the configured delay time (after:) another single group member will be called. If all members of a group are called, the process will be restarted with the first member of the group of the call order. In the first rotation the next member of the group of the call order will be called immediately if a user is busy. In the second rotation the busy users will hear knocking (only from external). The order of calls depends on the configured **Priority**: The first loggedin group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call.

Adding - fixed - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already

ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - balancing - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If a call was not accepted, although all logged-in members of the group were called, the call would be finished with a busy signal.

Adding - fixed - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends

on the configured **Priority**. If finally all logged-in members of a group are called, the busy users will hear knocking additionally (only from external).

Adding - balancing - rotating - Split Group

Here initially only one group members' phone is ringing. After the configured delay time (after:) another single group member will be called. If the group receives another call B, initially the free group members will ring one after the other (adding). If for the first call A less than half of the group members are ringing, more members are added to this call until exactly the half is reached. If for the first call A more than half of the users are ringing, no members are added to this call (frozen). Only the remaining users are added to call B. If all group members are already ringing for call A, the group is split for the calls. The order of calls depends on the configured **Priority**: The first logged-in group member gets the position 1 in the order of calls, the second one position 2, If position 1 takes a call, the second call will be assigned to position 2. The member of the group with the position 1 will get the last position in the order for a future call after he has finished the current call. If finally all loggedin members of a group are called, the busy users will hear knocking additionally (only from external).

Further help under Call Strategy

Exchange line settings

Property of the group

Opens the configuration of the single function for all groups.

Further help under **Groups**

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- Note: When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Transmit the own number

Defines the phone number of the assigned account as the phone number that will handle outgoing phone numbers.

Further help under Number presentation

Own number

Assigns an account, and therefore also a location, to the group. You can select the accounts that have been assigned to the group in the call distribution.

Further help under Number presentation

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Further help under Exchange line routing

Reachability

Property of the group

Opens the configuration of the single function for all groups.

Further help under Groups

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Reserved call channels

Number of reserved external VoIP channels for the group. The reservation can only be made on one account and the associated provider. These call channels are only available for users who are logged in this group as outgoing.

Further help under Reserved call channels

Busy-on-Busy

Callers are put through depending on the following factors:

- Number of available group members: The number is determined for each call. Incoming logged-in group members with at least one registered telephone are taken into account.
- Number of active connections: The number is made up of active connections of the group members and (if announcement before answering is enabled) callers who were automatically answered and connected to the waiting loop.

deactivated

Callers get busy as soon as the number of active connections equals the number of available group members.

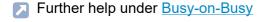
Exception: If announcement before answering is enabled, the reachability of the group and thus the calls connected to the waiting loop is unlimited.

dynamically

Callers get busy as soon as the number of active connections equals the number of available group members.

busy from x connections on

Callers get busy as soon as the number of connections set here is reached.



Announcement before answering

Off

When calling a group, no announcement is played. Available group members are immediately called. If all group members are busy, the caller hears the busy tone.

On busy

When calling a group, an announcement is played only if busy. Available group members are immediately called. If all group members are busy, an announcement is played for greeting. After the announcement, the caller is put on hold and hears Music on Hold.

Always

When calling a group the caller always hears the announcement for greeting. If group members are available, they are called. If all the group members are busy, the caller is put on hold after the announcement.

Further help under Announcement before answering

Announcement before answering | Call signalling

Immediately

The call is signalled during the announcement.

After Announcement before Answering

The call is signalled after the announcement when the caller hears Music on Hold.

Further help under Announcement before answering

Announcement before answering | Selection of announcement

Announcement which is played for the caller as greeting.

Further help under <u>Announcement before answering</u>

Rework time

Time that a group member cannot be called after an incoming internal or external call (e.g. for call rework). During this time, the Do-not-Disturb function is enabled for the user.

- Note: The rework time can be cancelled at any time by switching off the Do-not-Disturb function (e.g. using a function key at the system phone).
- **Note:** The following characteristics apply for the devices COMfortel 3200/3500: Enabling this function on the telephone is not displayed in the web interface of the PBX. Enabling this function in the web interface of the PBX has no influence on the telephone (status is not displayed under the function keys)

Blocklist | incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for group members logged in as outgoing.

Like user

The blocklists selected for the users are used.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Reachability

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Allowlist | incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for group members logged in as outgoing.

Like user

The allowlists selected for the users are used.

Existing allowlists

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Multi-path call forwarding | Destination number:

Internal and external calls to the group are signalled additionally to the entered destination number.

- Note: Multi-path Call Forwarding for a group is executed during the entire calling period independent of the call strategy for the group. It is not executed if the group itself is the destination of a forwarded call.
- Further help under Multi-path call forwarding

Call forwarding | Settings | only external calls

External calls to the group are forwarded. Internal calls are not forwarded.

Further help under Call forwarding

Call forwarding |Settings | CF on group call

Calls directed to a group of which the group is a member will be forwarded to the CF destination of the group.

Further help under Call forwarding

Call forwarding | CF immediately | Destination number:

Internal and external calls to the group are forwarded immediately to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on busy | Destination number:

Internal and external calls to the group are forwarded on busy to the entered destination number.

Further help under Call forwarding

Call forwarding | CF on no reply | Destination number:

Internal and external calls to the group are forwarded after the entered delay time to the entered destination number.

Delay time for CF on no reply

Call forwarding after the waiting time entered here in sec.

Further help under <u>Call forwarding</u>

Call forwarding | on all logged out | Destination number:

Internal and external calls to the group are forwarded to the entered destination number, if all group members are logged out.

Further help under Call forwarding

Voice mail and fax boxes

Property of the group

Opens the configuration of the single function for all groups.

Further help under Groups

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under **Profiles**

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Fax box | Telephone number of the box

A created fax box for the group is required in order to have the possibility of selecting a telephone number.

- Note: If it is not clear whether it is a fax or not, the call will be signalled to the group or it will be accepted by the voice mailbox.
- Note: If a fax call is accepted by the group, it can be forwarded to a fax box by means of transfer.

- Note: A fax call which is accepted by a voice mailbox will not be forwarded to a fax box.
- Further help under Voice mail and fax boxes

Voice mailbox | Telephone number of the box

Both the group and the selected voice mailbox will be automatically called for all incoming calls made to the internal number of the group.

- Note: If call forwarding immediately (user/group) is enabled, the call will not be made to the voice mailbox.
- Further help under Voice mail and fax boxes

Voice mailbox | Call acceptance type

immediately

The voice mailbox will be called immediately.

on no reply

The voice mailbox will be called after expiration of the entered delay time.

on busy

The voice mailbox will be called if the internal number of the group is busy and if no call forwarding on busy has been enabled for the group.

on no reply + on busy

The voice mailbox will be called if the group does not reply or if the internal number of the group is busy.

Further help under Voice mail and fax boxes

On all logged out

The voice mailbox will be called when all group members are logged out.

Further help under Voice mail and fax boxes

Call acceptance time on no reply

Call acceptance during the time entered here in sec.

Further help under Voice mail and fax boxes

Settings

Property of the group

Opens the configuration of the single function for all groups.

Further help under Groups

Property of the profile

Opens the configuration of the single function for all profiles.

Further help under Profiles

Profile-controlled

The profile setting overrides the group setting.

- **Note:** When switched off, the setting made for the group is valid.
- Further help under Profiles
- Further help under Groups

Number presentation (internally)

Group numbers

All group members logged in as outgoing will present the group number for internal outgoing calls. If the group is internally called, the user number is transferred back.

User phone number

Settings

All group members logged in as outgoing will present their own user number for internal outgoing calls.

Further help under Number presentation

Preliminary pick-up query and monitoring

Requirement is a VoIP end device which can evaluate the available information by BLF.

All information about the incoming calls of the group is available for evaluation by BLF.

- Note: When switched off, no phone numbers of the group's calls are transmitted by BLF.
- Further help under Preliminary pick-up query and monitoring

Pick-up of all calls to group members

A member of this group can pick up calls to the phone number of the group as well as direct calls to individual members of the group.

- **Note:** When switched off, group members can only pick up direct calls to the phone number of the group.
- Further help under Pick-up of all calls to group members

Voice mail and fax boxes

Phone numbers
Profiles
Functional Overview
Basic Settings

Phone numbers

Overview
Phone numbers
Configure

Overview

Memory utilisation free / total

The currently available and the totally available memory for voice mail and fax boxes in the PBX.

Further help under Voice mail and fax boxes

Phone numbers

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers.

- Further help under Internal numbers
- Further help under Voice mail and fax boxes

Name

Freely selectable name. This name is used to identify the box within the PBX and the connected VoIP phones.

Further help under Voice mail and fax boxes

Voice mailbox/Fax box

Select a voice mail or fax box.

Further help under Voice mail and fax boxes

Assignment

Select the internal number of a user or group that is to be assigned to the box.

An owner (user or group) can query the box. If the administrator has not refused the access authorisation, the owner can perform the following changes/actions using the web interface:

· Copying, moving, and deleting messages

- Adjustment of box properties which are not controlled via a profile under Features (except User/authorisations and Maximum recording capacity)
- Adjustment of properties of users and groups which are not controlled via a profile under Voice mail and fax boxes > Functional Overview

For a voice mailbox, the owner can additionally perform the following changes/actions:

- Recording, storing in the PBX, reading from the PBX (downloading), listening to and deleting of announcements
- Adjustment of box properties which are not controlled via a profile under Call acceptance/announcement

The following properties of the owner will be applied to the box:

- Exchange line routing of the user or group (for a group, the box has no exchange line authorisation with the setting Exchange line routing: like user)
- Note: Boxes can be preconfigured by not assigning an owner to the box. Nevertheless, the boxes cannot be used until an owner is assigned.
- Further help under Voice mail and fax boxes

Messages (saved / maximum)

Number of the stored and the maximum possible messages.

Further help under Voice mail and fax boxes

Properties (in use / maximum)

The maximum possible and already used by messages data storage in MB.

Further help under Voice mail and fax boxes

Configure

Opens the configuration of the voice mailbox or the fax box.

Details about settings under <u>Configure</u>

Q Filter

The list is filtered by the entered or selected value.

+ New

Depending on the selection, a new voice mailbox or fax box will be added. You must then make the settings for this box.

Several boxes with a contiguous number range can be created in the fields **from** | **until**.

Further help under Voice mail and fax boxes

Delete

Deletes the entries marked in the list.

Configure

Box settings
Users / Authorisations
Announcements
Call acceptance / Announcement

Box settings

Voice mail and fax boxes

Select the voice mail or fax box you want to display/configure.



Fax ID | Sender name

Enter a sender name that is to be inserted into the top line of a fax along with the fax ID

Further help under Voice mail and fax boxes

Fax ID | Phone number

Enter the complete phone number of the own external connection as fax ID. Number in the format +[country code][city code][external phone number] (e.g. +495306920094).

The fax ID is inserted into the top line of a fax during the fax transfer and, thus, permits an exact identification of the sender.

- Note: Recipients of a fax frequently send their responses to the number entered in the header of the fax. Therefore, the **Phone number** should be one of your own extensions/phone numbers with fax acceptance.
- Further help under Voice mail and fax boxes

Fax transmission | ECM (Error Correction Mode)

Transmission errors (e.g. because of faulty lines) will be corrected provided that ECM operation (Error Correction Mode) has also been configured for the opposite end.

- Note: If transmission errors occur very often (e.g. overseas lines of VoIP connections with frequent packet losses), the transmission cannot be carried out since the fax machines involved will cancel the transmission due to an excessive number of transmission repetitions. In such cases, it may be sensible to disable EMC error correction. Without ECM error correction, the transmission errors occurred will be displayed as erroneous dots or lines.
- Further help under Voice mail and fax boxes

Fax transmission | A maximum of 9600 baud

The transmission rate will be limited to 9600 baud and adapted in such a way that a faultless transmission quality is reached.

- Note: The PBX supports a maximum transmission rate of 14400 baud. This transmission rate may be too high for specific lines (e.g. overseas lines or VoIP connections) and may lead to transmission interferences. In such cases, a limitation of 9600 baud makes sense.
- Further help under Voice mail and fax boxes

Language of the announcements

If several language files have been imported, the language of the announcements can be selected.

Default

The configured standard language is used.

Further help under Voice mail and fax boxes

Automatic replay of new messages

All new messages will be replayed automatically without the need for further inputs and marked as listened to if the voice mailbox is queried or if message forwarding is used.

- Note: This setting does not affect querying the voice mailbox with a phone.
- Further help under Voice mail and fax boxes

Remote access

A voice mailbox can be queried from an external connection. The remote access can be performed by the administrator or owner/user after entering the corresponding PIN.

Further help under Voice mail and fax boxes

PIN for remote access

Freely selectable PIN. Helps to identify the permitted user when remotely querying the voice mail box.

- Note: If no entry is made, the PIN of the box owner is automatically used.
- Further help under Access authorisations

Message forwarding | Destination number

Incoming new messages are forwarded to the configured destination.

Destination number

Enter an internal or external phone number without exchange line access number.

Further help under Voice mail and fax boxes

Message forwarding | Forwarding after

Number of messages, after which the forwarding can be started (depending on the configured notification time period).

Further help under Voice mail and fax boxes

Message forwarding | Notification period

Time period in which the messages are to be forwarded.

Further help under Voice mail and fax boxes

Message forwarding | Number of repetitions

Attempts of call forwarding and pauses in between.

Further help under Voice mail and fax boxes

Call pick-up

The user can take over a call which was accepted by the voice mailbox.

global

Every internal user is permitted to take over calls.

Authorised users

Every owner or user is permitted to take over calls. If a group is owner or user, every group member can take over calls.

Further help under Voice mail and fax boxes

Maximum recording capacity (Voice mailbox)

Enter a value for the recording length between **15** up to **999**MB. Adjoining it is shown how many minutes the entered value is. If no value is entered, the recording capacity is unlimited (maximum memory utilisation).

Specifies the maximum memory capacity in MB available for the recording of messages.

- Note: Under the following circumstances, the entered value is not available for the box:
 - There is not enough memory capacity available on the storage device.
 - The memory capacity on the storage device is utilised by other boxes.

Maximum recording capacity (Fax box)

Enter a value for the recording length between **15** up to **999**MB. If no value is entered, the recording capacity is unlimited (maximum memory utilisation).

Specifies the maximum memory capacity in MB available for the fax box.

Box settings

- Note: Under the following circumstances, the entered value is not available for the box:
 - There is not enough memory capacity available on the storage device.
 - The memory capacity on the storage device is utilised by other boxes.
- Further help under Voice mail and fax boxes

Maximum length per recording

Enter a value for the recording length between **1** up to **99** min. If no value is entered, the recording length is unlimited (maximum memory utilisation).

Specifies the number of minutes after which the box ends a recording automatically (unless the caller already hung up).

- Note: Under the following circumstances, the entered value is not available for the box:
 - There is not enough memory capacity available on the storage device.
 - The memory capacity on the storage device is utilised by other boxes.
- Further help under Voice mail and fax boxes

Automatic clean-up

Perform at xx % memory utilisation

When the value (in the range of **50** up to **95** %) is reached, the oldest messages marked as read will be deleted as soon as new messages arrive. Archived messages will not be deleted.

- Note: If the clean-up fails, this information is logged and, if specified, sent as an e-mail.
- Further help under Voice mail and fax boxes

E-mail transfer | Send e-mail in HTML format

- Note: When switched off, e-mails are sent in text format.
- Further help under <u>E-mail</u>
- Further help under Voice mail and fax boxes

E-mail transfer | Send on incoming message (Voice mailbox)

Attach incoming message as file

In the case of an incoming message, this information is sent as an e-mail.

Audio file in MP3 format

The audio file is provided in MP3 format.

Note: When switched off, the audio file is available in wav format.

Mark as read the messages sent

In the message list, a sent message is marked as read.

E-mail transfer | Send on incoming message (Fax box)

Attach incoming message as file

Note: Faxes are provided as pdf files.

Mark as read the messages sent

- Important: If automatic marking as read is enabled, the fax box will be cleaned up automatically. This option should normally be enabled
- Further help under E-mail
- Further help under Voice mail and fax boxes

E-mail transfer | Send at xx % memory utilisation

Enter a value for the recording length between **50** up to **95** %. When the amount of memory used exceeds the specified level, this information is sent as an e-mail

- Further help under E-mail
- Further help under Voice mail and fax boxes

E-mail transfer | Send in the case of failure of automatic cleanup

If the automatic clean-up fails, an e-mail is sent. The settings made for the transfer of system e-mails will apply.

- Further help under <u>E-mail</u>
- Further help under <u>Voice mail and fax boxes</u>

E-mail transfer | Send after fax transfer

off

Disables e-mail transfer after fax transfer.

in the case of errors

Enables e-mail transfer after failed fax transfer.

Always

Enables e-mail transfer after fax transfer.

Attach sent message to file

The sent fax is attached as file to the e-mail.

- Further help under E-mail
- Further help under Voice mail and fax boxes

E-mail transfer | Outgoing mail server

Default

The e-mails are transferred via the default outgoing mail server.

Name of an outgoing mail server

The e-mails are transferred via the selected outgoing mail server.

- Further help under E-mail
- Further help under Voice mail and fax boxes

E-mail transfer | Language file

Select the language file.

- Note: If a user-defined language file is selected, e-mails will be sent with the text modules of the selected language file.
- Further help under E-mail
- Further help under Voice mail and fax boxes

E-mail transfer | Logo

Default logo

The default logo is used.

Name of user-defined logos

The selected logo is used.

- Further help under E-mail
- Further help under Voice mail and fax boxes

Message recipients

The e-mail addresses of up to 10 message recipients can be entered for e-mail transfer.

- Further help under E-mail
- Further help under Voice mail and fax boxes

▽ Test

By sending a test e-mail, it can be checked whether the e-mail transfer to the message recipients entered works without fault.

- Further help under E-mail
- Further help under Voice mail and fax boxes

Users / Authorisations

Voice mail and fax boxes

Select the voice mail or fax box you want to display/configure.

Further help under Voice mail and fax boxes

User

A box can only be used if an owner has been assigned to it (first entry in this overview). In addition to the owner, further users can be assigned to the box (if necessary, following entries in this overview).

Requirements:

- · Configured internal users and/or groups
- **Note:** An owner is automatically entered as a user with all authorisations.
- Note: All group members have the authorisations assigned to the group. If a group member or a subgroup is also entered individually as a user, the authorisations assigned to this user will also apply to this group member/subgroup.
- Further help under Voice mail and fax boxes

Administrate announcements

The authorisations of the announcements are expanded:

 Recording, storing in the PBX, reading from the PBX (downloading), listening to and deleting of announcements.

- Adjustment of properties which are not controlled via a profile under Call acceptance/announcement.
- Further help under Voice mail and fax boxes

Administrate settings

The authorisations of the properties of the box that are not controlled via a profile are expanded (except for **User/authorisations**, **Maximum recording capacity** and **Fax ID**).

- Note: Only a user may change these settings via telephone.
- Further help under Voice mail and fax boxes

+ New

Adds a new user/new group.

Further help under Voice mail and fax boxes

Announcements

Voice mail and fax boxes

Select the voice mail or fax box you want to display/configure.

Further help under Voice mail and fax boxes

Selection

Selection of an announcement place with or without recorded announcement.

Further help under Voice mail and fax boxes

Refresh

Refreshes the list entries according to import, recording, renaming or deleting of one or several announcement(s).

Further help under Voice mail and fax boxes

<u></u> **↓** Import

For each voice mailbox, a maximum of ten announcements with a maximum length of three minutes each can be saved in the PBX.

The audio file (*.wav) stored on a data storage device is selected and imported.

Note: The audio files (*.wav) must have the following properties: Format wideband (16-bit, 16 kHz, mono) and file name containing 32 characters.

Announcements

- Note: In the Internet, you can convert the audio files into the audio format required for the PBX (see <u>Auerswald Audio Converter</u>). For example, use the Audio Converter to convert audio files (*.mp3, *.mp4, *.wav or *.raw) into G.722 or G.711 (*.wav) audio format.
- Further help under Voice mail and fax boxes

∴Export

The announcements for the voice mailboxes saved in the PBX can be saved on the PC as backup copies.

Further help under Voice mail and fax boxes

Record

Announcements for voice mailboxes can directly be recorded via the receiver of an internal telephone.

As a user

The telephone is being called.

As an administrator

Select the phone via the list field **Record announcement at the user**. The telephone is being called.

The call must be accepted and after the tone an announcement can be recorded.

Further help under Voice mail and fax boxes

/ Edit

Changing the announcement name.

Further help under Voice mail and fax boxes

m Delete

The announcement recorded for the announcement place will be deleted. The announcement place remains in the list for a new recording.

Further help under Voice mail and fax boxes

Call acceptance / Announcement

Voice mail and fax boxes

Select the voice mail or fax box you want to display/configure.

Further help under Voice mail and fax boxes

Overridden by profile

The settings done are only valid as long as they are not overridden by a profile.

[Name configuration]

In the displayed configuration, the function is overridden by the displayed setting.

not overridden

The function is currently not overridden by a profile.

(not profile-controlled)

The function cannot be controlled with a profile.



Further help under Profiles

Readiness

On

Enables readiness.

off

Disables readiness.

Configuration-specific

The setting made in the profile assigned to the currently valid configuration will apply.

Further help under Voice mail and fax boxes

Reject anonymous fax calls

Incoming anonymous faxes (without transferred number) are not accepted.

Requirement is the enabled **Readiness**.

Further help under Voice mail and fax boxes

Substitute function

Requirements:

- · Corresponding announcement saved in the PBX
- Enabled Readiness of the voice mailbox

A caller can set up a connection to a substitute. For this purpose, the caller dials **1**

Destination number

Enter an internal or external phone number.

- Note: For an external number that has less than five digits, the local area code should additionally be entered.
- Further help under Voice mail and fax boxes

Announcement with enabled substitute function

Select the announcement to be played to callers if the substitute function is enabled.



Further help under Voice mail and fax boxes

Call acceptance / Announcement

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types. The Call acceptance/ announcement defines the standard behaviour for the case that no Callspecific behaviour applies.

Requirements:

- Enabled Readiness
- Disabled Substitute function
- If Box announcement: corresponding announcement saved in the **PBX**

No call acceptance

All calls not controlled via Call-specific behaviour will not be accepted. The caller hears the ringing tone.

Rejecting a call

All calls not controlled via Call-specific behaviour will not be accepted. The caller hears the busy signal (user/group will not be called additionally) or the call tone (user/group will be called).

Standard announcement

All calls not controlled via **Call-specific behaviour** will be accepted. The announcement of the selected language file will be replayed.

Box announcement

An announcement according to the selection in the alongside list field will be replayed.



Recording

The caller can leave a message. For this, additionally the option **Save short messages** must be switched on for another outgoing mail server.

Further help under Voice mail and fax boxes

Save short messages

The phone number of the caller is displayed even when the caller does not leave a message or hangs up during the announcement. This setting also applies to **Call-specific behaviour**.

Further help under Voice mail and fax boxes

Call-specific behaviour (exception table)

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types.

Note: If the call-specific behaviour is permanently set in a configuration by means of a profile, the corresponding table can be viewed. Click the corresponding profile name which is displayed in red font. The corresponding profile configuration is displayed.

Requirements:

- Enabled Readiness
- Disabled Substitute function
- If Box announcement: corresponding announcement saved in the PBX
- Further help under Voice mail and fax boxes

Call type

Internal calls

Call acceptance for internal calls.

Door calls

Call acceptance for door terminal calls.

External calls

Call acceptance for external calls.

Anonymous callers

Call acceptance for calls without transferred telephone number.

Custom

Call acceptance for calls from the entered phone number or phone number range entered under **Ext. anonymous callers**.



Ext. anonymous callers

Entering for call acceptance via **Call type**: **Custom**.

Possible is either the complete external phone number or a phone number range starting for example with 00 for international calls.



Further help under Voice mail and fax boxes

Call acceptance

Default

For call acceptance see Call acceptance/announcement under Standard behaviour.

No call acceptance

No reaction on calls, the caller hears the ringing tone.

Rejecting a call

The caller hears the busy signal (user/group will not be called additionally) or the call tone (user/group will be called).

Standard announcement

The announcement of the selected language file will be replayed.

Box announcement

The setting in the Announcements list field selected announcement of the voice mailbox will be replayed.



Further help under Voice mail and fax boxes

Recording

The caller can leave a message. For this, additionally the option Save **short messages** must be switched on for another outgoing mail server.



Profiles

Profiles

Profile management
Profile Assignment
Configure

Profile management

Name

The available profiles (default profiles, user-defined profiles) are displayed.

Further help under Profiles

View

The selected settings can be viewed (default profiles).

Further help under Profiles

Configure

The user-defined profiles can be adapted to the corresponding requirements (user-defined profiles).

Details about settings under <u>Configure</u>

+New

Creation of a new profile.

Further help under Profiles

Profile Assignment

Assignment

The selected profile is assigned to the user, the group or the box.



All

The same profile is assigned to all users, groups or boxes.

Further help under Profiles

Configure

Configure

Call acceptance / Announcement

Call acceptance / Announcement

Profile Configuration

Select the profile you want to display/configure.

Further help under Profiles

Accept profile settings from

Select the profil whose settings are to be taken.

Further help under Profiles

Profile-controlled settings

The profile setting overrides the voice mail or fax box setting.

- Note: When switched off, the setting made for the voice mail or fax box is valid.
- Further help under <u>Profiles</u>

Readiness

On

Enables readiness.

off

Disables readiness.

Configuration-specific

The setting made in the profile assigned to the currently valid configuration will apply.

Further help under Voice mail and fax boxes

Reject anonymous fax calls

Incoming anonymous faxes (without transferred number) are not accepted.

Requirement is the enabled **Readiness**.

Further help under Voice mail and fax boxes

Substitute function

Requirements:

- Corresponding announcement saved in the PBX
- Enabled Readiness of the voice mailbox

A caller can set up a connection to a substitute. For this purpose, the caller dials **1**.

Destination number

Enter an internal or external phone number.

- Note: For an external number that has less than five digits, the local area code should additionally be entered.
- Further help under Voice mail and fax boxes

Announcement with enabled substitute function

Select the announcement to be played to callers if the substitute function is enabled.



Further help under Voice mail and fax boxes

Call acceptance / Announcement

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types. The Call acceptance/ announcement defines the standard behaviour for the case that no Callspecific behaviour applies.

Requirements:

- Enabled Readiness
- Disabled Substitute function
- If Box announcement: corresponding announcement saved in the **PBX**

No call acceptance

All calls not controlled via Call-specific behaviour will not be accepted. The caller hears the ringing tone.

Rejecting a call

All calls not controlled via Call-specific behaviour will not be accepted. The caller hears the busy signal (user/group will not be called additionally) or the call tone (user/group will be called).

Standard announcement

All calls not controlled via **Call-specific behaviour** will be accepted. The announcement of the selected language file will be replayed.

Box announcement

An announcement according to the selection in the alongside list field will be replayed.



Recording

The caller can leave a message. For this, additionally the option **Save short messages** must be switched on for another outgoing mail server.

Further help under Voice mail and fax boxes

Save short messages

The phone number of the caller is displayed even when the caller does not leave a message or hangs up during the announcement. This setting also applies to **Call-specific behaviour**.

Further help under Voice mail and fax boxes

Call-specific behaviour (exception table)

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types.

Note: If the call-specific behaviour is permanently set in a configuration by means of a profile, the corresponding table can be viewed. Click the corresponding profile name which is displayed in red font. The corresponding profile configuration is displayed.

Requirements:

- Enabled Readiness
- Disabled Substitute function
- If Box announcement: corresponding announcement saved in the PBX
- Further help under Voice mail and fax boxes

Call type

Internal calls

Call acceptance for internal calls.

Door calls

Call acceptance for door terminal calls.

External calls

Call acceptance for external calls.

Anonymous callers

Call acceptance for calls without transferred telephone number.

Custom

Call acceptance for calls from the entered phone number or phone number range entered under **Ext. anonymous callers**.



Ext. anonymous callers

Entering for call acceptance via **Call type**: **Custom**.

Possible is either the complete external phone number or a phone number range starting for example with **00** for international calls.



Call acceptance

Default

For call acceptance see **Call acceptance/announcement** under **Standard behaviour**.

No call acceptance

No reaction on calls, the caller hears the ringing tone.

Rejecting a call

The caller hears the busy signal (user/group will not be called additionally) or the call tone (user/group will be called).

Standard announcement

The announcement of the selected language file will be replayed.

Box announcement

The setting in the **Announcements** list field selected announcement of the voice mailbox will be replayed.



Recording

The caller can leave a message. For this, additionally the option **Save short messages** must be switched on for another outgoing mail server.

Functional Overview

Call acceptance / Announcement

Call acceptance / Announcement

Property of the box

Opens the configuration of the single function for all voice mail and fax boxes.

Property of the profile

Opens the configuration of the single function for all profiles.

Profile-controlled settings

The profile setting overrides the voice mail or fax box setting.

Note: When switched off, the setting made for the voice mail or fax box is valid.

Readiness

On

Enables readiness.

off

Disables readiness.

Configuration-specific

The setting made in the profile assigned to the currently valid configuration will apply.

Reject anonymous fax calls

Incoming anonymous faxes (without transferred number) are not accepted.

Requirement is the enabled **Readiness**.

Further help under Voice mail and fax boxes

Substitute function

Requirements:

- Corresponding announcement saved in the PBX
- Enabled Readiness of the voice mailbox

A caller can set up a connection to a substitute. For this purpose, the caller dials **1**

Destination number

Enter an internal or external phone number.

- Note: For an external number that has less than five digits, the local area code should additionally be entered.
- Further help under Voice mail and fax boxes

Announcement with enabled substitute function

Select the announcement to be played to callers if the substitute function is enabled.

Call acceptance / Announcement

The call acceptance of the voice mailbox can be configured to show different behaviour for different call types. The **Call acceptance/ announcement** defines the standard behaviour for the case that no **Call-specific behaviour** applies.

Requirements:

- Enabled Readiness
- Disabled Substitute function
- If Box announcement: corresponding announcement saved in the PBX

No call acceptance

All calls not controlled via **Call-specific behaviour** will not be accepted. The caller hears the ringing tone.

Rejecting a call

All calls not controlled via **Call-specific behaviour** will not be accepted. The caller hears the busy signal (user/group will not be called additionally) or the call tone (user/group will be called).

Standard announcement

All calls not controlled via **Call-specific behaviour** will be accepted. The announcement of the selected language file will be replayed.

Box announcement

An announcement according to the selection in the alongside list field will be replayed.

Recording

The caller can leave a message. For this, additionally the option Save short messages must be switched on for another outgoing mail server.



Further help under Voice mail and fax boxes

Save short messages

The phone number of the caller is displayed even when the caller does not leave a message or hangs up during the announcement. This setting also applies to Call-specific behaviour.



Basic Settings

Basic settings for all voice mail and fax boxes Language files

Basic settings for all voice mail and fax boxes

System Information | Generate notification at xx % memory utilisation

When the amount of memory used exceeds the specified level, this information is logged and, if specified, sent as an e-mail. The settings made for the transfer of system e-mails will apply.

- Further help under Voice mail and fax boxes

Automatic clean-up | Perform at xx % memory utilisation

When the amount of memory used reaches the specified level, the oldest message that is marked as read is deleted when a new message is received. Archived messages will not be deleted.

- Note: If the clean-up fails, this information is logged and, if specified, sent as an e-mail.
- Further help under Voice mail and fax boxes

Automatic clean-up | Type of clean-up

oldest messages in the system

The oldest messages marked as read of all messages stored in the voice mail/fax system will be deleted (even if all messages belong only to one box).

oldest messages in each box

The oldest messages of each box marked as read will be deleted.

Basic settings for all voice mail and fax boxes

Further help under Voice mail and fax boxes

Memory | Automatic memory check

The data storage device is automatically checked for errors at the selected interval and at the selected time.

- Note: If the memory check fails, this information is logged and, if specified, sent as an e-mail.
- Further help under Voice mail and fax boxes

Memory | Interval

daily

The memory check will be made daily.

weekly

The memory check will be made weekly.

monthly

The memory check will be made on the 1st day of every month.

Further help under Voice mail and fax boxes

Memory | Time

Time at which the memory check is performed.

Note: The memory check may take a few minutes. The voice mail and fax boxes cannot be accessed during the memory check. For this reason, the memory check should be performed at a time when access is not required (e.g. at night).

Further help under Voice mail and fax boxes

Channels simultaneously used for outgoing calls

Maximum number of call channels that can be used simultaneously for outgoing calls from the Voice Mail/Fax function (message forwarding, fax transfer).



Language files

Default language

Unless a different language has been selected for specific voice mailboxes, the selected language is usually used for all the voice mailbox announcements.

- Note: The first language imported into the PBX is automatically set as Default language.
- Further help under Voice mail and fax boxes

丛 Import

The language file (*.fs) stored on a PC in fs format is selected and imported.

The voice mailboxes' default announcements, and also the spoken user guidance for remote access, are stored in "language files" (for Englishlanguage announcements, english.fs is used). Up to four language files can be stored simultaneously. This means different languages can be used for different voice mailboxes.

Further help under Voice mail and fax boxes

C Refresh

The overview of the language files is refreshed.

Telephones

Assignment Status Provisioning

Assignment

Overview Configure Adding

Overview

Telephone type

Select the telephone type or DECT base you want to display/configure.

Further help under <u>Hardware</u>

Name

Freely selectable name. This name is used to identify the device within the PRX

Further help under <u>Hardware</u>

MAC address

The MAC address of the device is used for identification. For all transactions between device and PBX, the PBX must be able to identify the MAC address of the device so that the device can be matched to provisioning data.

The MAC address of a device can be found as follows:

- · Retrieving from a client certificate sent by the device
- Reading with a bar code scanner enabled and connected to the computer (e.g. via a USB interface)
- Reading e.g. on the menu of the phone or on a label
- Note: If the MAC address was determined via automatic provisioning or entered by yourselves, the device will be provided with provisioning data during its first boot process.
- Further help under <u>Hardware</u>

Manufacturer

The manufacturer name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under <u>Hardware</u>

Model

The model name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under Hardware

Version

The currently used firmware version for this device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.



Further help under Hardware

User

The user assigned to this telephone or the user currently logged onto this telephone.

┌ ---

No user is assigned or no user is logged on.

The telephone is freely available for users with roaming user authorisation.

A user is assigned or a user is logged on.

Currently not available for other users who have roaming user authorisation.

The current user can log out with roaming user authorisation or free up the telephone.

A user is permanently assigned. Therefore, he is the owner. The owner is logged on.

Not available for other users who have roaming user authorisation.

The owner can log out without any other authorisation, but remains the owner.

A user is permanently assigned. Therefore, he is the owner. The owner is logged out.

Not available for other users who have roaming user authorisation.

The owner can log on without any other authorisation.

Phone number | [name]

Mobile telephone A user is permanently assigned. Therefore, he is the owner. Incoming calls to the owner are signalled on the mobile telephone.

The owner can switch off calls to their mobile telephone without any other authorisation.

_ --- [(Phone number] | [name)]

Overview

Mobile telephone A user is permanently assigned. Therefore, he is the owner. Incoming calls to the owner are not signalled on the mobile telephone.

The owner can switch on calls to their mobile telephone without any other authorisation.



Mobile telephone No user assigned.

[Phone number] | [name]

Mobile telephone A user is assigned. Incoming calls to the user are signalled on the mobile telephone.

- Note: If no assignment is possible, the maximum number of enabled users has been exceeded. Another user licence is necessary. The number of used/available user licences is displayed here: **Telephones** > **Assignment**.
- Further help under <u>User</u>
- Further help under Roaming User
- Further help under FMC

Configure

Opens the configuration of the device.

Details about settings under <u>Configure</u>

Filter

The list is filtered by the entered or selected value.

+ New

Adds a new telephone type/new DECT base.

Details about settings under <u>Adding</u>

m Delete

Deletes the entries marked in the list.

Configure

Device

Select the device you want to display/configure.

Further help under <u>Hardware</u>

Name

Freely selectable name. This name is used to identify the device within the PBX.

Further help under <u>Hardware</u>

SIP user name

Displays the phone's SIP user name. This consists of the internal phone number of the user assigned to the telephone and another digit, if the user is assigned to several telephones. (Exception: SIP soft client.)

Further help under VolP

Address of the provisioning server

Displays the address of the used provisioning server.

Further help under Provisioning

Password

Automatically generated, freely changeable password. Used for SIP registration.

- Note: To change the current password, click on

 Edit PIN and password. After you have entered your PIN or password, click on

 Hide PIN and password.
- Further help under <u>Access authorisations</u>

Sends the access data for registering the Softphone to the entered e-mail address.

Generate new password

A new password will be generated before the access data is sent.

- Further help under Softphone
- Further help under <u>E-mail</u>

Login PIN

Handset login PIN.

Further help under DECT system

CTI access | Token

Freely selectable authorisation code. Requirement is a telephone of the COMfortel D series (firmwaree version 1.4 or later).

- Note: To change the current password, click on Edit PIN and password. After you have entered your PIN or password, click on Hide PIN and password.
- Further help under LAN TAPI

Manufacturer

The manufacturer name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under Hardware

Model

The model name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under <u>Hardware</u>

Version

The currently used firmware version for this device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Configure

Further help under <u>Hardware</u>

MAC address

The MAC address of the device is used for identification. For all transactions between device and PBX, the PBX must be able to identify the MAC address of the device so that the device can be matched to provisioning data.

The MAC address of a device can be found as follows:

- Retrieving from a client certificate sent by the device
- Reading with a bar code scanner enabled and connected to the computer (e.g. via a USB interface)
- Reading e.g. on the menu of the phone or on a label
- Note: If the MAC address was determined via automatic provisioning or entered by yourselves, the device will be provided with provisioning data during its first boot process.
- Further help under Hardware

Provisioning file

Provisioning file assigned to the device.

Further help under Provisioning

Device IPUI

Selection of an unassigned IPUI or manual entry for a handset COMfortel M-5x0/M-7x0.

Configure

The IPUI can be determined as follows:

- Read out the barcode label on the package with a barcode reader connected to the computer (e.g. via the USB interface)
- Reading from the handset: Press the control button and enter *#06#
- · Reading from the package label

To identify a handset, the IPUI of the handset is used. For all transactions between the handset, base station and PBX, the PBX must be able to determine the IPUI of the handset.

Further help under DECT system

COMfortel WS-500x base station

The handset COMfortel M-5x0/M-7x0 is assigned to a base station here.

Further help under DECT system

Location

Location at which the telephone is located. The correctly assigned location is necessary so that the PBX can properly initiate an emergency call. The account of the user who is currently assigned to the telephone is used to determine the location for all other calls.

- Further help under Locations
- Further help under **Emergency calls**

Jitter buffer

Specifies the size of the jitter buffer in milliseconds (duration of the audio signals). In this time period RTP packets are cached (lower values = lower number, greater values = greater number). Delays in the transmission and loss of packets can be compensated that way.



Further help under VolP

Prevent early update messages

Disables early SIP update messages, e.g. in the event of connection establishment problems.



Further help under VolP

SIP alert info

Shows whether the telephone can evaluate the specified SIP alert info. If problems occur with signalling on the telephone, switch off the option involved

Auto Answer Calls

Enables calls to be received automatically. Basic requirement for reacting to InterCom.

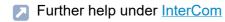
External calls

Distinguishes between external calls and all other calls. Ringtones can be assigned to calls if the telephone supports this functionality.

Internal, external, group and door terminal calls

Configure

Distinguishes between the types of call made. Ringtones can be assigned to calls if the telephone supports this functionality.



RTP encryption

Allows an exception from the default settings of the PBX for the corresponding VoIP user.

Settings are required in the following cases:

- The corresponding device does not support the settings of the PBX. This may cause problems e.g. during a call being established.
- You must bypass the PBX settings for testing or troubleshooting purposes.
- Note: The settings of the PBX for the corresponding user and the settings of the device must match.

Default (Administration > VoIP > Internal SIPS/SRTP)

The PBX setting applies for the user.

Only encrypted

Example: SRTP is disabled in the PBX. The device requires an enabled SRTP

Only unencrypted

Example: SRTP is enabled in the PBX. The device does not support SRTP.

Further help under VolP

Format of transferred telephone number

The PBX supports several formats to display the country code of incoming calls on the display of the VoIP phone being called, e.g. adjusted to a small display.

- Note: With the telephones COMfortel 3200/3500 and COMfortel D series, the format country code with leading + sign (e.g. + 49 5306) cannot be used.
- Further help under Number presentation

Features of the phone

If an IP telephone is registered on the PBX and the IP address is known, use this button to reach the web interface of the device. There you can make further settings.

Further help under <u>Hardware</u>

Module

Module to which the device is to be connected.

Further help under Hardware

Port

Port to which the device is to be connected. Only the available ports are displayed.

Further help under Hardware

CLIP info

The phone number information (if transmitted) is displayed on the display of the device.

Requirement is a CLIP-capable analogue device.

Further help under <u>Phone number display (CLIP)</u>

CLIP info extended

In the caller list of the device date and time of missed calls are recorded. The name can be displayed instead of or in addition to the phone number.

Requirement is a CLIP-capable analogue device.

- Note: For some devices, extended CLIP information can cause problems and should therefore be disabled.
- Further help under Phone number display (CLIP)

Hook flash timing

Hook flash timing enables the PBX to detect a flash signal from the device in question.

For this to work, the hook flash timing must correspond with the time that is configured for the flash signal on the analogue end device (see the device manual).

Note: DTMF phones that execute various functions using the T-Net function key/menu use a flash of 300 ms to do this. In order to also use these functions on the PBX, the same time as the hook flash timing must be configured on the PBX. Usually, there is an additional Flash

Configure

key available on this phone that must also be configured with this value (see the phone manual).

- Note: Note that the PBX does not detect a receiver being hung up until after hook flash timing is finished.
- Further help under Query and Transfer

Duration of the 1st ringing signal

Specifies how many milliseconds the 1st ringing signal lasts. Under certain circumstances, this setting can influence the CLIP function of analogue telephones.

0 ms

The 1st ringing signal is suppressed. Default value for United Kingdom location.

1000 ms

The 1st ringing signal lasts 1000 milliseconds. Default value for other locations.

Further help under Hardware

Mobile telephone number

Phone number of the mobile telephone used for FMC.

Further help under FMC

Adding

Name

Freely selectable name. This name is used to identify the device within the PBX.



Further help under <u>Hardware</u>

Type

Type of telephone or DECT base to be configured.

SIP telephone

SIP telephone. The requirement here is that provisioning from the PBX must be supported.

SIP soft client. Manufacturer: Auerswald

COMfortel SoftPhone

PBX Call Assist

SIP soft client. Manufacturer: Another manufacturer

SIP soft client or SIP telephone on which provisioning cannot be performed

DECT base

COMfortel WS-500S or WS-500M

DECT handset

Handset COMfortel M-5x0/M-7x0 on a base station COMfortel WS-500S or WS-500M

Mobile telephone

Mobile telephone used for FMC.

Analogue telephone

Telephone connected to the internal analogue port. The prerequisite for this is a COMtrexx Business with plugged-in 4FXS module.

- Further help under Provisioning
- Further help under Softphone
- Further help under <u>DECT system</u>
- Further help under FMC
- Further help under <u>Hardware</u>

Manufacturer

The manufacturer name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under Hardware

Model

The model name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under <u>Hardware</u>

Version

The currently used firmware version for this device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under <u>Hardware</u>

MAC address

The MAC address of the device is used for identification. For all transactions between device and PBX, the PBX must be able to identify the MAC address of the device so that the device can be matched to provisioning data.

The MAC address of a device can be found as follows:

- · Retrieving from a client certificate sent by the device
- Reading with a bar code scanner enabled and connected to the computer (e.g. via a USB interface)
- Reading e.g. on the menu of the phone or on a label
- Note: If the MAC address was determined via automatic provisioning or entered by yourselves, the device will be provided with provisioning data during its first boot process.
- Further help under <u>Hardware</u>

Device IPUI

Selection of an unassigned IPUI or manual entry for a handset COMfortel M-5x0/M-7x0.

The IPUI can be determined as follows:

- Read out the barcode label on the package with a barcode reader connected to the computer (e.g. via the USB interface)
- Reading from the handset: Press the control button and enter *#06#
- Reading from the package label

To identify a handset, the IPUI of the handset is used. For all transactions between the handset, base station and PBX, the PBX must be able to determine the IPUI of the handset.

Further help under DECT system

Login PIN

Handset login PIN.

Further help under DECT system

COMfortel WS-500x base station

The handset COMfortel M-5x0/M-7x0 is assigned to a base station here.

Further help under DECT system

Provisioning file

Provisioning file assigned to the device.

Further help under Provisioning

Module

Module to which the device is to be connected.

Further help under <u>Hardware</u>

Port

Port to which the device is to be connected. Only the available ports are displayed.

Further help under <u>Hardware</u>

Mobile telephone number

Phone number of the mobile telephone used for FMC.

Further help under FMC

User

Assigns a user to the telephone.

- Note: If no assignment is possible, the maximum number of enabled users has been exceeded. Another user licence is necessary. The number of used/available user licences is displayed here: **Telephones** > **Assignment**.
- Further help under User

Send registration link

Sends the access data for registering the Softphone to the entered e-mail address.

- Further help under <u>Softphone</u>
- Further help under E-mail

Location

Location at which the telephone is located. The correctly assigned location is necessary so that the PBX can properly initiate an emergency call. The account of the user who is currently assigned to the telephone is used to determine the location for all other calls.

- Further help under Locations
- Further help under **Emergency calls**

Status

Telephone type

Select the telephone type or DECT base you want to display/configure.

Further help under <u>Hardware</u>

Name

Freely selectable name. This name is used to identify the device within the PRX

Further help under Hardware

MAC address

The MAC address of the device is used for identification. For all transactions between device and PBX, the PBX must be able to identify the MAC address of the device so that the device can be matched to provisioning data.

The MAC address of a device can be found as follows:

- · Retrieving from a client certificate sent by the device
- Reading with a bar code scanner enabled and connected to the computer (e.g. via a USB interface)
- Reading e.g. on the menu of the phone or on a label
- Note: If the MAC address was determined via automatic provisioning or entered by yourselves, the device will be provided with provisioning data during its first boot process.
- Further help under <u>Hardware</u>

Manufacturer

The manufacturer name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under <u>Hardware</u>

Model

The model name of the device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under Hardware

Version

The currently used firmware version for this device. If automatic provisioning is enabled, the name is entered by the PBX as long as the device provides this information.

Further help under Hardware

User

No user has been assigned to this telephone.

[Phone number] | [name]

A user has been assigned to this telephone. The user can be permanently assigned (owner) or not.

--- [(Phone number] | [name)]

A temporarily assigned user (owner) is logged out at this time.

Further help under <u>Hardware</u>

Status

Display of registration (IP phones), connection (analogue phones) and provisioning of devices (telephones, DECT bases, door terminals) and (not yet) assigned devices by different symbols. These are explained in the following.

- Note: When calling the page, the current status is displayed first and is then automatically refreshed after 30 seconds.
- Note: Not all status information can be updated automatically, see following sections.
- Further help under <u>Hardware</u>

Additional information

Information about the individual devices (IP/MAC address, model name, firmware version, status).

Further help under <u>Hardware</u>

Provisioning

Settings of the provisioning server

Opens the configuration of the provisioning server.

Details about settings under <u>Provisioning</u>

Name

File name of the provisioning file.

Further help under Provisioning

Device type

Select the device to be configured.

A template is available for each device type supprted by the PBX. These files cannot be changed, deleted or exported.

Further help under Provisioning

Editor

Via the file editor, settings (e.g. NTP or contacts) are directly made in the provisioning file (*.xml, *.cfg).

Certain predefined files (e.g. standard.xml) are not editable and deletable in the integrated file editor but can be looked at.

/ Edit

The provisioning files (*.xml, *.cfg) can be edited using the editor.

Variables for the parameters to be provisioned are created as placeholders in the configuration files provided by the manufacturer. During provisioning, they are replaced by the corresponding values in the PBX configuration.

- Note: We recommend to contact the system administrator or specialised dealer for editing provisioning files.
- Note: Under Account IDs, you find the account IDs for the PBX and the exchange lines.
- Further help in the <u>Auerswald DokuWiki</u>.
- Further help under Provisioning

+ New

Creating a new provisioning file and selection of the device type this provisioning file is to be used for.

Further help under Provisioning

m Delete

Deletes the entries marked in the list.

Further help under <u>Provisioning</u>

Import

Import of an existing provisioning file in the formats *.xml and *.cfg.

Further help under <u>Templates</u>

∴ Export

Export of one or more provisioning file(s) from the PBX as backup copy on the PC.

- Note: When exporting more than one provisioning file at the same time: mark all desired files before the export. They are saved in a provisioning.zip folder.
- Further help under Provisioning

Internal Devices

Automatic receptions
Conference rooms
Call on hold

Automatic receptions

Overview Configure

Overview

Internal number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by an internal user.

- Further help under <u>Internal numbers</u>
- Further help under <u>Automatic reception</u>

Name

Freely selectable name. This name is used to identify the automatic reception within the PBX.

Further help under <u>Automatic reception</u>

Configure

Opens the configuration of the automatic reception.

- Details about settings under Configure
- Further help under <u>Automatic reception</u>

+ New

Creates a new entry.



Deletes the entries marked in the list.

Configure

Internal number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for distributing calls and can be called directly by an internal user.

- Further help under <u>Internal numbers</u>
- Further help under <u>Automatic reception</u>

Name

Freely selectable name. This name is used to identify the automatic reception within the PBX.

Further help under <u>Automatic reception</u>

Announcement

Select the greeting to be played to callers. Make your selection from the announcements for automatic receptions and the text before answering announcements.

- Note: Under some circumstances, a greeting with background music may cause errors because the audible touch tones cannot be properly recognized, or parts of the music may be interpreted as touch tones.
- Further help under <u>Automatic reception</u>

Max. number of external calls on hold in parallel

Specifies how many external calls can be kept on hold at the same time. The caller hears the busy signal if this number is exceeded. External calls on hold are calls that have either not yet been transferred or calls that have been transferred back to the automatic reception. This restriction does not apply to internal callers and to callers transferred by a superordinated reception.



Further help under Automatic reception

Max. number of the repetitions per call

Specifies how often the greeting is played if the caller does not input a valid number. The system reacts as if the destination was busy/did not reply, after the announcements and the subsequent waiting time has elapsed.



Further help under Automatic reception

Waiting time between the announcement or until transfer to the default destination

Specifies how long the PBX waits (in seconds) for a valid input between and after the announcements.



Further help under <u>Automatic reception</u>

Reaction on busy/on no response of the destination

If an invalid number is redialled or if the destination is busy/does not reply, the announcement if the destination is busy/does not reply is played. The system then reacts according to the following setting.

End of the call

Three short busy signals are played and the PBX ends the existing call with the caller.

Call default destination

The PBX transfers the existing call to the default destination number entered below. If the call is not answered there either, three short busy signals are played and the PBX ends the existing call.

If the destination number and the default destination number are identical, the PBX cuts off the call without calling again.

Repeat Automatic Reception Final amount

The PBX transfers the existing call to the automatic reception that was called previously.



Further help under Automatic reception

Announcement on busy/on no reply of the destination

Select the announcement to be played to a caller if the destination is busy or does not reply. Make your selection from the announcements for automatic receptions and the text before answering announcements, or do not select an announcement.



Further help under <u>Automatic reception</u>

Default destination number

Internal phone number the call is transferred to, if no destination has been chosen or optional, if the destination is not available.

The following internal destinations are possible:

User

Configure

- Group
- Voice mailbox
- · Different automatic reception
- Note: An external destination can be reached by entering an internal destination that is forwarded to an external one (e.g. via call forwarding).
- Further help under <u>Automatic reception</u>

Music on Hold during the ringing period

If the Music on hold during the ringing period option has been enabled, a caller transferred by the automatic reception will only hear the music on hold during this process.

- Note: When switched off, the caller hears the ringtone during the ringing phase.
- Further help under <u>Automatic reception</u>

DTMF | Destination number

Internal phone number to be transferred to after dialling the DTMF suffix dialling digits 0-9.

Alternative value for DTMF 0-9:*. This value allows the caller the targeted redial of an internal number within the relevant phone number range such as for DTMF 4 phone numbers starting with 4.

The following internal destinations are possible:

- User
- Group

Configure

- Voice mailbox
- · Different automatic reception
- Note: An external destination can be reached by entering an internal destination that is forwarded to an external one (e.g. via call forwarding).
- Further help under <u>Automatic reception</u>

Conference rooms

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for direct calling by an internal user

- Further help under Internal numbers
- Further help under Conference rooms

Name

Freely selectable name. This name is used to identify the conference room within the PBX.

Further help under <u>Conference rooms</u>

Announcement

Announcement which is played to a conference participant when entering the conference room.

Further help under Conference rooms

+ New

Creates a new entry.

m Delete

Deletes the entries marked in the list

Call on hold

Internal basis telephone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. When put together with the ten possible park positions 0 - 9, this creates ten 3 to 4-digit phone numbers. These numbers can then be used to select the parking zone for call parking.

- Further help under <u>Internal numbers</u>
- Further help under <u>Call on hold</u>

Name

Freely selectable name. This name is used to identify the internal number within the PBX.

Further help under <u>Call on hold</u>

Hold time

Specifies the maximum minutes a call is kept in the internal parking zone. After the hold time has elapsed, the user that put the call in the parking zone is again signalled that the call is still there. If this user is busy, the call is put back on hold in the parking zone. If the hold time elapses a second time without the call being transferred, the connection is disconnected and the caller hears the busy signal.

Further help under <u>Call on hold</u>

External Devices

Fax
Device wizard
Door stations
Announcement outputs
Relays (actuators)

Fax

Fax

Overview Configure

Overview

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers.

- Further help under Internal numbers
- Further help under Fax

Name

Freely selectable name. This name is used to identify the fax machine within the PBX.

Further help under <u>Fax</u>

Module

Module to which the device is connected.

Further help under <u>Hardware</u>

Port

Port to which the device is connected.

Further help under <u>Hardware</u>

Configure

Opens the configuration of the fax machine.

Overview

Details about settings under <u>Configure</u>

Q Filter

The list is filtered by the entered or selected value.

+ New

Creates a new entry.

Further help under <u>Fax</u>

m Delete

Deletes the entries marked in the list.

Configure

Blocklist | Incoming

A caller, who transmitted one of the phone numbers entered in the selected blocklist or no number, is rejected.

None

No blocklist is used for the device.

Existing blocklists

Selection of an already created blocklist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under Block- and allowlist

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Allowlist | Incoming

The Do-not-Disturb function does not apply to callers who use one of the phone numbers entered in the selected allowlist.

None

No allowlist is used for the device.

Existing allowlists

Configure

Selection of an already created allowlist.

Edit | Block- and allowlists

Opens the overview and the configuration of all block- and allowlists.

Further help under <u>Block- and allowlist</u>

Configure

Opens the configuration of the block- or allowlist.

Details about settings under <u>Configure</u>

Transmit the own number

Assigns an account, and therefore also a location, to the device. You can select the accounts that have been assigned to the device in the call distribution.

Defines the phone number of the assigned account as the phone number that will handle outgoing phone numbers.

Further help under Number presentation

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Further help under Exchange line routing

CLIP info

The phone number information (if transmitted) is displayed on the display of the device.

Requirement is a CLIP-capable analogue device.

Further help under Phone number display (CLIP)

CLIP info extended

In the caller list of the device date and time of missed calls are recorded. The name can be displayed instead of or in addition to the phone number.

Requirement is a CLIP-capable analogue device.

- Note: For some devices, extended CLIP information can cause problems and should therefore be disabled.
- Further help under Phone number display (CLIP)

Hook flash timing

Hook flash timing enables the PBX to detect a flash signal from the device in question.

For this to work, the hook flash timing must correspond with the time that is configured for the flash signal on the analogue end device (see the device manual).

Note: DTMF phones that execute various functions using the T-Net function key/menu use a flash of 300 ms to do this. In order to also use these functions on the PBX, the same time as the hook flash timing must be configured on the PBX. Usually, there is an additional Flash

Configure

key available on this phone that must also be configured with this value (see the phone manual).

- Note: Note that the PBX does not detect a receiver being hung up until after hook flash timing is finished.
- Further help under **Query and Transfer**

Duration of the 1st ringing signal

Specifies how many milliseconds the 1st ringing signal lasts. Under certain circumstances, this setting can influence the CLIP function of analogue telephones.

0 ms

The 1st ringing signal is suppressed. Default value for United Kingdom location.

1000 ms

The 1st ringing signal lasts 1000 milliseconds. Default value for other locations.

Further help under <u>Hardware</u>

Device wizard

Overview Configure

Overview

Name

Freely selectable name. This name is used to identify the device within the PBX.

Device template

Select the device template corresponding to the device.

Further help under <u>Templates</u>

Module

Module to which the device is connected.

Further help under Hardware

Port

Port to which the device is connected.

Further help under <u>Hardware</u>

Configure

Opens the configuration of the device template.

Details about settings under <u>Configure</u>

+ New

Creates a new device template

Further help under <u>Templates</u>

Delete

Deletes the entries marked in the list.

Imports a new device template:

- · from a file in .xml format or
- in form of a configuration provided by the manufacturer
- Further help under **Templates**

File management

Lists all device templates.

Further help under **Templates**

Configure

Analogue Door Terminals
VoIP Door Terminals
IP switch relay
Announcement outputs

Analogue Door Terminals

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number can be called directly by an internal user. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers.

- Further help under <u>Internal numbers</u>
- Further help under <u>Door functions</u>

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

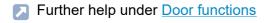
Switch input | Name

Enter the names of the relays and switch inputs.

Further help under <u>Door functions</u>

Destination number

Select or enter the required call destination (external number, internal number of the individual phone, internal number of the group).



DTMF sequence

Enter the request that is assigned to the door bell button within the door terminal (possible entries: Digits and characters *, #).

Further help under **Door functions**

X Close

The dialogue for the configuration is closed, already created data is discarded.

Save and close

The dialogue for the configuration is closed, already created data is saved.

Displays further entry fields for configuring additional parameters.

VolP Door Terminals

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number can be called directly by an internal user. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers.

- Further help under <u>Internal numbers</u>
- Further help under <u>Door functions</u>

SIP password

SIP password required by the connected device.

IP address

Enter the IP address required by the connected device. Possible entries: IP address with a range of 0 to 255 (example: 192.168.0.240).

Port

Enter the port required by the connected device. Possible entries: 1 to 65535.

User name

Enter the user name (authentication data to control the relays).

Password

Password (authentication data to control the relays).

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

Switch input | Name

Enter the names of the relays and switch inputs.

Further help under **Door functions**

Destination number

Select or enter the required call destination (external number, internal number of the individual phone, internal number of the group).

Further help under <u>Door functions</u>

SIP invite

Enter the SIP invites which the connected door terminal uses as signal. (Possible entries: Digits and the characters * and #).

VoIP Door Terminals

Further help under **Door functions**

× Close

The dialogue for the configuration is closed, already created data is discarded.

Save and close

The dialogue for the configuration is closed, already created data is saved.

∞ Expert

Displays further entry fields for configuring additional parameters.

IP switch relay

IP address

Enter the IP address required by the connected device. Possible entries: IP address with a range of 0 to 255 (example: 192.168.0.240).

Port

Enter the port required by the connected device. Possible entries: 1 to 65535.

User name

Enter the user name (authentication data to control the relays).

Password

Password (authentication data to control the relays).

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

IP switch relay

× Close

The dialogue for the configuration is closed, already created data is discarded.

Save and close

The dialogue for the configuration is closed, already created data is saved.

∞ Expert

Displays further entry fields for configuring additional parameters.

Announcement outputs

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for direct calling by an internal user.

- Further help under Internal numbers
- Further help under <u>Loudspeaker Audio Out</u>

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

× Close

The dialogue for the configuration is closed, already created data is discarded.

Save and close

The dialogue for the configuration is closed, already created data is saved.

∞ Expert

Displays further entry fields for configuring additional parameters.

Door stations

Overview
Configuration of the door terminal
Configuration of the relay
Configuration of the door call

Overview

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number can be called directly by an internal user. When put together with the numbers of the door bell buttons, this creates 3 to 4-digit phone numbers.

- Further help under <u>Internal numbers</u>
- Further help under <u>Door functions</u>

Name

Freely selectable name. This name is used to identify the device within the PBX.

Further help under <u>Door functions</u>

Module | Port

Module and port to which the device is connected.

Further help under <u>Hardware</u>

Configure

Opens the configuration of the door terminal.

Details about settings under <u>Configuration of the door terminal</u>

Relays | + New

Press to open a list for selecting an already configured relay.

Further help under Relays (actuators)

Relays | • Overview

Changes for further settings to the menu Relays (actuators).

Further help under Relays (actuators)

Relays | 前 Delete

Deletes the relay.

Relays | / Configure

Opens the configuration of the relay.

Details about settings under <u>Configuration of the relay</u>

Door calls | + New

Press to open a list for selecting from already configured relays or for adding a new one.

Further help under <u>Door functions</u>

Door calls | m Delete

Deletes the door call.

Door calls | / Configure

Opens the configuration of the door call.

- Details about settings under <u>Configuration of the door call</u>
- Further help under **Door functions**

Configuration of the door terminal

Configuration of the door terminal

Select the door terminal to be configured.



Further help under Door functions

Enable connection

Settings of the DTMF sequences for controlling the door terminal to enable the connection. The DTMF sequences used by the connected device are to be found in the according instruction. You can use the comma ',' character as a pause of half a second.



Further help under Door functions

Finish connection

Settings of the DTMF sequences for controlling the door terminal to terminate the connection. The DTMF sequences used by the connected device are to be found in the according instruction. You can use the comma ',' character as a pause of half a second.



Further help under **Door functions**

Dead time

Waiting time in milliseconds before the PBX sends DTMF sequences to the door terminal again. Possible entries: 0 to 10000 milliseconds. Time settings for receiving DTMF sequences again are to be made in accordance with the instructions of the connected devices.

Configuration of the door terminal

Further help under **Door functions**

SIP password

SIP password required by the connected device.

Further help under **Door functions**

X Close

The dialogue for the configuration is closed, already created data is discarded.

□ Save and close

The dialogue for the configuration is closed, already created data is saved.

Configuration of the relay

Configuration of the relay

Select the relay to be configured.

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Select an operating mode.

Further help under Relays (actuators)

PBX is decisive (primary)

Available relays can be toggled without affecting the PBX. After a reboot of the PBX the last identifiable state of the relay is restored.

Further help under Relays (actuators)

Off control

Swichtes off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration. When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Further help under Relays (actuators)

Switching duration

Specifies after how many seconds the relay will be switched off. Possible entries: 0 to 6000 (0 only for manual relays). It can only be switched off after the set switching duration when off control is enabled.

Further help under Relays (actuators)

Hold time

Specifies how many milliseconds the relay is held. Possible entries: 10 to 10000. The hold time is used for the relays whose switching off cannot be controlled anymore. A sufficiently long enought hold time will make sense. A relay is only held for the set time when off control is disabled.

Further help under Relays (actuators)

Dead time

Waiting time in milliseconds before the PBX sends DTMF sequences to the door terminal again. Possible entries: 0 to 10000 milliseconds. Time settings for receiving DTMF sequences again are to be made in accordance with the instructions of the connected devices.

Further help under Relays (actuators)

Switching | Switch on

Settings of the required DTMF sequences in order to switch on an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Switching | Switch off

Settings of the required DTMF sequences in order to switch off an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Close

The dialogue for the configuration is closed, already created data is discarded

Save and close

The dialogue for the configuration is closed, already created data is saved.

Configuration of the door call

Configuration of the door call

Select the door call to be configured.

Further help under **Door functions**

DTMF sequence

Enter the request that is assigned to the door bell button within the door terminal (possible entries: Digits and characters *, #).

Further help under **Door functions**

SIP invite

Enter the SIP invites which the connected door terminal uses as signal. (Possible entries: Digits and the characters * and #).

Further help under **Door functions**

Tabs

Configuration of different configurations.

Further help under <u>Configuration switchover</u>

Destination number

Select or enter the required call destination (external number, internal number of the individual phone, internal number of the group).

Further help under **Door functions**

Call duration

Select a call duration during which a door call is to be signalled to an internal user. The maximum call duration set in the connected door terminal will be limited by the PBX.

Further help under **Door functions**

Exchange line dialling / number presentation | DDI

Specific selection of the external connection and the number to be transmitted with it

All ext. lines or internal All fields must be filled out!

Door calls can be made on internal or on all external channels.

External Connection

Door calls will be made on the selected external connection.

DDI

Entry of the DDI to be transmitted from the extensions block.



Further help under Door functions

Second ringer

A second ringer can additionally be installed at locations where ringing of the phone will not always be heard. It will be triggered by a relay with the operating mode second ringer which is assigned to the door call.



Further help under **Door functions**

Announcement outputs

Overview
Configuration of the announcement output
Configuration of the relay

Overview

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers. This number is then used for direct calling by an internal user

- Further help under Internal numbers
- Further help under <u>Loudspeaker Audio Out</u>

Name

Freely selectable name. This name is used to identify the device within the PBX.

Further help under <u>Loudspeaker Audio Out</u>

Module | Port

Module and port to which the device useable as announcement output is connected.

Further help under <u>Hardware</u>

Relays

Overview of the entered parameters for announcement outputs.

Further help under Loudspeaker Audio Out

Configure

Opens the configuration of the announcement output.

Details about settings under <u>Configuration of the announcement output</u>

Relays | + New

Press to open a list for selecting an already configured relay.

Further help under Relays (actuators)

Relays | Noverview

Changes for further settings to the menu Relays (actuators).

Further help under Relays (actuators)

Relays | 前 Delete

Deletes the relay.

Relays | / Configure

Opens the configuration of the relay.

Details about settings under <u>Configuration of the relay</u>

Configuration of the announcement output

Configuration of the announcement output

Select the announcement output to be configured.



Further help under Loudspeaker Audio Out

Enable connection

Settings of the DTMF sequences for controlling the announcement output to enable the connection. The DTMF sequences used by the connected device are to be found in the according instruction. You can use the comma ',' character as a pause of half a second.



Further help under Loudspeaker Audio Out

Finish connection

Settings of the DTMF sequences for controlling the announcement output to terminate the connection. The DTMF sequences used by the connected device are to be found in the according instruction. You can use the comma ',' character as a pause of half a second.



Further help under Loudspeaker Audio Out

Dead time

Waiting time in milliseconds before the PBX sends DTMF sequences to the announcemebt output again. Possible entries: 0 to 10000 milliseconds. Time settings for receiving DTMF sequences again are to be made in accordance with the instructions of the connected devices.

Configuration of the announcement output

Further help under Loudspeaker Audio Out

SIP password

SIP password required by the connected device.

Further help under Loudspeaker Audio Out

X Close

The dialogue for the configuration is closed, already created data is discarded.

□ Save and close

The dialogue for the configuration is closed, already created data is saved.

Configuration of the relay

Configuration of the relay

Select the relay to be configured.

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

PBX is decisive (primary)

Available relays can be toggled without affecting the PBX. After a reboot of the PBX the last identifiable state of the relay is restored.

Further help under Relays (actuators)

Off control

Swichtes off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration. When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Further help under Relays (actuators)

Switching duration

Specifies after how many seconds the relay will be switched off. Possible entries: 0 to 6000 (0 only for manual relays). It can only be switched off after the set switching duration when off control is enabled.

Further help under Relays (actuators)

Hold time

Specifies how many milliseconds the relay is held. Possible entries: 10 to 10000. The hold time is used for the relays whose switching off cannot be controlled anymore. A sufficiently long enought hold time will make sense. A relay is only held for the set time when off control is disabled.

Further help under Relays (actuators)

Dead time

Waiting time in milliseconds before the PBX sends DTMF sequences to the door terminal again. Possible entries: 0 to 10000 milliseconds. Time settings for receiving DTMF sequences again are to be made in accordance with the instructions of the connected devices.

Further help under Relays (actuators)

Switching | Switch on

Settings of the required DTMF sequences in order to switch on an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Switching | Switch off

Settings of the required DTMF sequences in order to switch off an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Close

The dialogue for the configuration is closed, already created data is discarded

Save and close

The dialogue for the configuration is closed, already created data is saved.

Relays (actuators)

Relays (actuators)

Overview Configuration of the relay

Overview

Device name

Device name.

Configure

Opens the configuration of the device template.

Details about settings under <u>Configure</u>

no.

Number of the relay. The number is automatically assigned when a new relay is created.

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

IdentNumber

Identification number of the relay. Identification number is automatically assigned when a new relay is created.

Further help under Relays (actuators)

Module

Module to which the device is connected

Overview

Further help under Hardware

Port

Port to which the device is connected.

Further help under <u>Hardware</u>

Operating mode

Select an operating mode.

Further help under Relays (actuators)

used by /

Assigns a device to the relay.

Further help under Relays (actuators)

Status

Displays the switching status of the device.

Further help under Relays (actuators)

Configure

Opens the configuration of the device template.

Details about settings under <u>Configuration of the relay</u>

+ New

Creates a new entry.

Further help under Relays (actuators)

m Delete

Deletes the entries marked in the list.

Configuration of the relay

Configuration of the relay

Displays the relay to be configured.

Relays | Name

Name of the relay as under Relays (actuators).

Further help under Relays (actuators)

Operating mode

Displays the selected operating mode.

Further help under Relays (actuators)

PBX is decisive (primary)

Available relays can be toggled without affecting the PBX. After a reboot of the PBX the last identifiable state of the relay is restored.

Further help under Relays (actuators)

Off control

Swichtes off relays with the operating modes Light, Door opener, Alarm and PC relay after the specified switching duration. When switched off, the hold time of the relay will be used. The PBX does not actively switch off the relay.

Further help under Relays (actuators)

Switching duration

Specifies after how many seconds the relay will be switched off. Possible entries: 0 to 6000 (0 only for manual relays). It can only be switched off after the set switching duration when off control is enabled.

Further help under Relays (actuators)

Hold time

Specifies how many milliseconds the relay is held. Possible entries: 10 to 10000. The hold time is used for the relays whose switching off cannot be controlled anymore. A sufficiently long enought hold time will make sense. A relay is only held for the set time when off control is disabled.

Further help under Relays (actuators)

Dead time

Waiting time in milliseconds before the PBX sends DTMF sequences to the door terminal again. Possible entries: 0 to 10000 milliseconds. Time settings for receiving DTMF sequences again are to be made in accordance with the instructions of the connected devices.

Further help under Relays (actuators)

Switching | Switch on

Settings of the required DTMF sequences in order to switch on an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Switching | Switch off

Settings of the required DTMF sequences in order to switch off an analogue relay. Possible entries: Digits and the characters * and # and the comma ',' character as a pause of half a second.

Further help under Relays (actuators)

Close

The dialogue for the configuration is closed, already created data is discarded

Save and close

The dialogue for the configuration is closed, already created data is saved.

Functions

Time control
Call Through
CLIP texts
Short-code number
Music on Hold/announcements

Time control

Configurations
Switching times
Holidays
Overview

Configurations

Automatic switching Configurations Adding Copy

Automatic switching

Automatic configuration switching

At least two configured configurations are toggled according to the PBX internal clock.

- Note: If the automatic configuration switchover is enabled, the configuration enabled under **Time control** > **Switching times** for the current time becomes active. If a different configuration is then selected manually, this remains active until the next switching time.
- Further help under <u>Configuration switchover</u>

Controllable with system relay: Aut.Konfig.um. (900)

Automatic configuration switchover is controllable via phone.

Further help under Configuration switchover

Configurations

Name

Freely selectable name. This name is used to identify the configuration within the PBX.

Further help under Configuration switchover

Identification number

Automatically issued number. Is used for manual switching of a configuration by a programming sequence.

Further help under <u>Configuration switchover</u>

Enable

The selected configuration is switched on. To switch off a configuration, switch on a different configuration.

Further help under Configuration switchover

+New

Adds a new configuration to the list. After selection of **Save**, the dialogue for the configuration opens.

Details about settings under <u>Adding</u>

Delete

Deletes the entries marked in the list

Copy

Configuration settings are copied from one configuration into other, selected configurations.



Details about settings under <u>Copy</u>

Adding

Take over settings from configuration

Select the configuration to be copied with its corresponding settings.

Further help under Configuration switchover

Configuration-dependent settings

Select the functions that are to be assigned to the newly added configuration and switched with it.

Further help under Configuration switchover

Copy

Copying a configuration

Select the configuration to be copied with its corresponding settings.

Further help under Configuration switchover

In configurations

Select the configuration(s) that shall get the settings of the configuration to be copied.

Further help under Configuration switchover

Configuration-dependent settings

Select the functions that are to be assigned to the newly added configuration and switched with it.

Further help under Configuration switchover

Switching times

Time

Time (format hh:mm) when the selected configuration alongside is to be activated.

- Note: A configuration applies until another configuration (manual or automatic) is enabled, also beyond the daily change. An exception is the transition from a holiday to a normal weekday. In this case, the configuration activated for the holiday will only be active until 23:59. The normal configuration for a weekday will be activated at 0:00 of the following day (even if a configuration was activated manually on the holiday).
- Further help under <u>Configuration switchover</u>

Configuration

Configuration to be activated.

Further help under <u>Configuration switchover</u>

+ New

Creates a new entry.

Further help under <u>Configuration switchover</u>

Delete

Deletes the entries marked in the list.

—Сору

Switching times are copied from one weekday to other selected weekdays.



Holidays

Holidays

Calendar Adding

Calendar

New

For individual data (e.g. company vacations). General holidays are added with **Insert list of holidays** or alternatively by double-clicking a day.

Details about settings under <u>Adding</u>

Deleting calendar

All holidays entered into the calendar will be deleted.

Further help under <u>Configuration switchover</u>

Import

Entered holidays of a PBX of the **COMtrexx** series that were previoulsy exported and saved on a data storage device in *.ft format, are imported here.

Further help under Configuration switchover

Export

The entered holidays are saved to the (PC) data storage device in an *.ft file so they can be imported afterwards to a different PBX of the **COMtrexx** series.

Further help under Configuration switchover

Insert list of holidays

The holidays valid for the country/federal state are added for an indefinite period. Holidays with yearly changing dates are calculated automatically by the PBX.

- Note: Holidays inserted in this manner can be edited or deleted individually as needed. Each of these changes is carried out automatically for all calendar years.
- Further help under Configuration switchover

Adding

Name

Freely selectable name.

Further help under Configuration switchover

Date from

First day of entry (e.g. company vacations).

Further help under Configuration switchover

Date until

Last day of entry (e.g. company vacations).

Further help under Configuration switchover

active

Entry is enabled as holiday.

Further help under Configuration switchover

once

Entry is enabled as holiday only for the current year.

Further help under Configuration switchover

Overview

Week

Displays the time periods of the automatically controlled configurations (12 a.m.-11:59 p.m.) as well as the manually enabled configurations.

Further help under Configuration switchover

Call Through

Exchange line settings Call Through user

Exchange line settings

PIN for Call Through

Freely selectable PIN. Helps to identify the transferred phone number as well as the permitted Call Through user.

- Note: To change the existing PIN, click on
 Edit PIN. After you have entered your PIN or password, click on
 Hide PIN.
- Further help under <u>Access authorisations</u>

Max. duration of external conversations

Specifies after how many minutes an external Call Through call is ended by the PBX, due to security reasons.

Further help under Call Through

Prevention of emergency calls

Emergency calls cannot be made. The Call Through user hears a corresponding announcement.

This function is useful as the emergency call is issued at the location of the PBX and not at the location of the Call Through user.

Further help under Emergency calls

Exchange line routing (outgoing)

Defines by selecting a rule under which conditions exchange line calls are performed.

Further help under Exchange line routing

Phone numbers for Call Through

External phone numbers, that the Call Through user must dial to call the PBX for Call Through. This is determined in the exchange line routing.

Further help under **Exchange line routing**

Call Through user

Display name

Select a previously created contact according to its display name.

Further help under Call Through

Phone number

Displays the phone numbers configured for this contact. The Call Through user must transfer one of these phone numbers for identification.

Further help under Call Through

Block- and allowlist

If block- and allowlist for the exchange line routing are configured, the phone number dialled by the Call Through user is checked.

Further help under <u>Block- and allowlist</u>

Filter

The list is filtered by the entered or selected value.

+New

Adds a new Call Through user to the list. Only configured Call Through users are allowed to perform this function.

Further help under Call Through

CLIP texts

TAPI

The configured CLIP text is displayed on analogue phones during a call that signals the LAN-TAPI function.

Further help under Phone number display (CLIP)

Callbacks

The configured CLIP text is displayed on analogue phones during a callback (on busy or on no reply).

Further help under Phone number display (CLIP)

Default

All user-defined settings will be reset to the standard values.

Further help under Phone number display (CLIP)

Short-code number

Phone number

Internal number, freely selectable from the numbers in the internal supply of phone numbers.

- Further help under Internal numbers
- Further help under **Short-code number**

Name

Freely selectable name.

Further help under **Short-code number**

Macro sequence

Simple digit sequence (e.g. ##06) or sequence of macro commands that are selected/triggered by the short-code.

Possible entries and their effect:

Note: Entry of external number without exchange line access number, regardless of the exchange line request

Simple digit sequence consisting of 0-9, * and #

Dialing a digit sequence

Macro command Z + digit sequence consisting of 0-9, * and #

Dialing a digit sequence

At the beginning of a sequence of macro commands: call start

Short-code number

Within a sequence of macro commands: DTMF transmission, provided there is a connection

Macro command W + digit 1-5

Within a sequence of macro commands: Waiting time/pause time of 1-5 seconds

Macro command C

Within a sequence of macro commands: Waiting for connection

Macro command H

At the end of a sequence of macro commands: Hang up

Further help under **Short-code number**

+ New

Creates a new entry.

Further help under **Short-code number**

Delete

Deletes the entries marked in the list.

Music on Hold/announcements

File management File assignment

File management

Memory utilisation
Existing audio files (*.wav)

Memory utilisation

Utilized

Memory already used for audio files (+.wav).

Maximum

The combined total size of all audio files stored in the PBX must be less than 20 MB.

Available

The memory still available for audio files in MB.

Remaining capacity

The memory still available for audio files in percentage terms.

Existing audio files (*.wav)

Own Files
System Files

Own Files

Source of own files

This list is where you manage the audio files you have created or specially obtained for the different PBX functions. You can for example use an external programme to create the audio files and then import them.

Note: Using audio files that have not been created by yourselves, take care that no copyrights are infringed.

Alternatively, you can record the announcements on one of the PBX's internal phones. You can then assign the following file names to the audio files you create for the different functions:

- AudioIn_MoH.wav (music on hold)
- Phone_Auto_x.wav (announcement for automatic reception)
- Phone_Hold_Line.wav (announcement for music on hold)
- Phone_TvM_x.wav (announcement for announcement before answering)
- emergency_call_not_possible_de.wav (announcement for emergency call)
- conference_room_announcex.wav (announcement for conference rooms)

Name

The file name of the audio file in format *.wav.

Further help under Music on Hold and Announcement

Dimensions

The size of the audio file in kB.

Further help under Music on Hold and Announcement

Length

The length of the audio file in format hh:mm:ss.

Further help under <u>Music on Hold and Announcement</u>

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

m Delete

Deletes the entries marked in the list.

丛 Import

The audio file (*.wav) stored on a data storage device is selected and imported.

Note: The audio files (*.wav) must have the following properties: Format wideband (16-bit, 16 kHz, mono) and file name containing 32 characters.

Own Files

- Note: In the Internet, you can convert the audio files into the audio format required for the PBX (see <u>Auerswald Audio Converter</u>). For example, use the Audio Converter to convert audio files (*.mp3, *.mp4, *.wav or *.raw) into G.722 or G.711 (*.wav) audio format.
- Further help under Music on Hold and Announcement

System Files

Standard assignment

The audio files already stored in the PBX for various different PBX functions are displayed in this list. The **Standard assignment** column shows which audio file is usually used for each function.

- Further help under <u>Music on Hold and Announcement</u>
- Further help under Automatic reception
- Further help under <u>Emergency calls</u>
- Further help under Conference rooms
- Further help under <u>Announcement before answering</u>

Name

The file name of the audio file in format *.wav.

Further help under Music on Hold and Announcement

Dimensions

The size of the audio file in kB.

Further help under <u>Music on Hold and Announcement</u>

Length

The length of the audio file in format hh:mm:ss.

Further help under Music on Hold and Announcement

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

File assignment

Music on hold
Announcement before answering
Automatic receptions
Conference Room

Music on hold

Music on hold with announcement

Music on Hold also during the ringing period

Music on hold

Music on hold with announcement

Record music on hold

The assigned music on hold is played.

Further help under Music on Hold and Announcement

Volume of music on hold during the announcement

Specifies how loudly the music on hold should be played during the announcement.

Further help under Music on Hold and Announcement

Mix announcement into Music on Hold

An announcement is played together with music on hold.

- **Note:** When switched off, music on hold is played without an additional announcement.
- Further help under Music on Hold and Announcement

Delay time of the announcement

Specifies how many seconds delay there will be between the individual repetitions of the announcement.

Further help under Music on Hold and Announcement

Music on Hold also during the ringing period

Transferring a call without an announcement

A caller connected by transferring without announcement will exclusively hear music on hold during this process (pressing the **R key**, dialling the internal number, and ringing period).

- Note: When switched off, the caller hears the ringtone during the ringing phase.
- Further help under Music on Hold and Announcement
- Further help under Query and Transfer

Announcement before answering

A caller hears the music on hold after the playback of the announcement before answering.

When switched off, the caller hears the ringtone during the ringing phase.

- Further help under Music on Hold and Announcement
- Further help under Announcement before answering

Music on hold

Function

Function for which the adjoining audio file is to be used.



Audio file (*.wav)

Select the audio file that is to be used for the adjoining function.

Standard announcement

For the standard announcement, the PBX uses the audio file specified as the standard file in Functions > Music on Hold/announcements > File management > System Files.

Further help under Music on Hold and Announcement

Volume on telephone

Specifies in percentage terms how loudly the audio file is to be played on the phone.

- Note: The volume at the phone for emergency calls cannot be set under 50 %.
- Further help under <u>Music on Hold and Announcement</u>

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

Announcement before answering

Function

Function for which the adjoining audio file is to be used.

- Further help under Music on Hold and Announcement
- Further help under <u>Announcement before answering</u>

Audio file (*.wav)

Select the audio file that is to be used for the adjoining function.

Standard announcement

For the standard announcement, the PBX uses the audio file specified as the standard file in Functions > Music on Hold/announcements > File management > System Files.

Further help under Music on Hold and Announcement

Volume on telephone

Specifies in percentage terms how loudly the audio file is to be played on the phone.

- Note: The volume at the phone for emergency calls cannot be set under 50 %.
- Further help under Music on Hold and Announcement

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

Automatic receptions

Function

Function for which the adjoining audio file is to be used.

- Further help under Music on Hold and Announcement
- Further help under <u>Automatic reception</u>

Audio file (*.wav)

Select the audio file that is to be used for the adjoining function.

Standard announcement

For the standard announcement, the PBX uses the audio file specified as the standard file in Functions > Music on Hold/announcements > File management > System Files.

Further help under Music on Hold and Announcement

Volume on telephone

Specifies in percentage terms how loudly the audio file is to be played on the phone.

- Note: The volume at the phone for emergency calls cannot be set under 50 %.
- Further help under Music on Hold and Announcement

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

Conference Room

Function

Function for which the adjoining audio file is to be used.

- Further help under Music on Hold and Announcement
- Further help under <u>Conference rooms</u>

Audio file (*.wav)

Select the audio file that is to be used for the adjoining function.

Standard announcement

For the standard announcement, the PBX uses the audio file specified as the standard file in Functions > Music on Hold/announcements > File management > System Files.

Further help under Music on Hold and Announcement

Volume on telephone

Specifies in percentage terms how loudly the audio file is to be played on the phone.

- Note: The volume at the phone for emergency calls cannot be set under 50 %.
- Further help under Music on Hold and Announcement

Listen

A suitable programme is used to play the audio file. Depending on the used programme, the file can then, if necessary, be stored on a data storage device.

Further help under Music on Hold and Announcement

User data

User data

Call data
Contacts and assignment
Messages

Call data

Call data
Print options
Filter
Entry

Call data

Selected filter

Filters the list of call data depending on the selected categories. **Standard filter** or **Separate filter**.

Further help under Call data

Date / time

Display of beginning of the call.

Further help under Call data

Duration

Display of the call duration.

Further help under <u>Call data</u>

External partner

Displays the phone number of the external call partner.

- Dialled calls: Displays the dialled phone number.
- **Incoming Calls**: Displays the phone number transmitted by the caller (CLIP information; not for unknown phone numbers).
- Important: Phone numbers of an incoming VoIP call can only be used for callbacks, if the Evaluation of incoming SIP messages is enabled.

- The phone number is entered in a blocklist.
- Name and phone number of the caller is transferred to the contacts.
- Further help under <u>Call data</u>

External name

Displays the name of the external call partner if transmitted via CLIP or stored in the PBX as contact.

As long as there is no entry for the VoIP phone number in the contacts, the name of the provider given in the PBX is entered.

Further help under <u>Call data</u>

No. invoice

Displays the internal user or group number for:

- Outgoing call: user phone number of the end device. If the internal user is logged in to a group, the group number is displayed.
- Incoming call: Internal user or group number dialled by the caller
- Call Through call: External number of the Call Through user
- Door-to-external calls: Phone number of the door terminal
- Further help under Call data

Name invoice

Displays the user or group name for:

Call data

- Internal user numbers
- Internal group numbers
- · Call Through user
- Door-to-external calls
- Further help under <u>Call data</u>

No. real

Displays the internal user or group number for:

- Outgoing call: Internal number of the executing end device.
- **Incoming call**: Internal number of the accepting end device or external number of the accepting external call forwarding destination (unsuccessful calls are not displayed).
- Door-to-external calls: Internal user number who forwarded the call to an external connection (direct calls to external users are not displayed).
- Further help under Call data

Name real

Displays the user or group name for:

- · Internal user numbers
- Internal group numbers
- · Call Through users or external numbers
- Door-to-external calls
- Further help under Call data

Connection

Displays the own phone number.

- Outgoing call: phone number that has been transmitted by the PBX
- Incoming call: Phone number dialled by the caller
- Call made via analogue connection: Analogue number
- Call via VoIP: VoIP number
- Further help under <u>Call data</u>

Direction

Outgoing or incoming call.

Further help under Call data

Type of billing

successful Successful call

unsuccessful Unsuccessful call

Further help under <u>Call data</u>

Call type

The text entered under **Print options** for the **Call type** is displayed.

Standard sorting

The call data is sorted by date and time.

Further help under Call data

C Refresh

The call data list is refreshed.

Further help under Call data

Export

The current data in the table can be exported (gespr_dat.csv) for further processing.

Important: Note the following legal notes before saving the call data.

The call data includes, amongst others, the following legally especially protected components:

- The time of a call
- The phone numbers of the internal and external participants.
- The duration of a call

Before the call data are being saved, all users of the PBX must be informed that these contents will be forwarded and about who will use the data received.

It is essential to store the data securely and in accordance with the legal requirements.

call data available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.

Further help under Call data

m Delete all

All displayed entries with the current filter will be deleted.

Print options

Table columns

The columns ordered as they are available under the page Call data.

Further help under Call data

Call data list

Display or hide the available columns on the page Call data.

Further help under Call data

Description in call data list

Displays the description of the column in the call data list (as **Default** or individually assigned).

Further help under Call data

Default

All settings made will be reset to the standard values.

Filter

Filter

Overview Configure

Overview

Filter name

Name of the used filter type (for standard filter) or freely assigned name (for specially configured filters).

Further help under Call data

Period

dynamically

All default filters or sepcially created filters such as filter type **current** week. Excluded from specially created filters is the filter type **User-defined**.

These filters adjust (always based on the current date) the pre-defined period of a filter type (e.g. **current week**).

static

Only for specially created filters as filter type **User-defined**.

These filters have an assigned period that is not dynamically adjusted with the current date.

Further help under <u>Call data</u>

The selected settings can be viewed (default filters).

Configure

Opens the configuration of an own filter.

Details about settings under <u>Configure</u>

+ New

Adds a new filter.

Based on

Select a standard filter and its proper settings for applying them on the new filter.

Further help under Call data

Delete

Deletes the entries marked in the list.

Configure

Period
Direction
Type of billing
Call type
Specific filtering

Period

Default period

Select a pre-defined period from the list.

Further help under Call data

Period

When selecting the (static) option **User-defined** from the **Default period** list: Enter own values

When selecting all other (dynamic) options from the **Default period** list: Enter the corresponding values in the fields. When selecting **current week**, **current month** or **current year**, the second entry field for the date is filled with the corresponding current date.

The time period of the dynamic options displayed is constantly adjusted to the current date.

Further help under <u>Call data</u>

Daytime

Enter the day time to be filtered.

Further help under Call data

Call duration

Enter the conversations with corresponding length to be filtered.

Direction

Incoming calls

Conversations resulting from incoming calls.

Further help under Call data

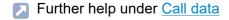
Outgoing calls

Conversations resulting from outgoing calls.

Type of billing

Successful Calls

Calls that resulted in a connected call.



Unsuccessful Calls

Calls that did not result in a connected call.

Call type

Characteristic a call may have

Select for filtering a certain type of calls or conversations.



Specific filtering

User (no. invoice + no. real)

Calls made by selected internal users.

Further help under Call data

Connection number

Calls initiated via the own external number.

Further help under Call data

External partner

Calls made with selected external call partners.

Further help under Call data

Anonymous callers

If enabled, caller that have the number presentation suppressed are also displayed.

Entry

Outgoing calls

Records all outgoing calls. By disabling one or several of the adjoining options, the amount can be limited, also to prevent the call data memory from filling up too quickly.

Further help under Call data

Chargeable calls (also unsuccessful)

Chargeable calls and unsuccessful outgoing calls (e.g. transfer charges).

Further help under Call data

Free of charge / calls without charge information

Calls that are free of charge or for which no charge information is transmitted.

Further help under Call data

Successful only

Calls that are free of charge or for which no charge information is transmitted (unsuccessful calls are not displayed).

Incoming calls

Records all incoming calls. By disabling one or several of the adjoining options, the amount can be limited, also to prevent the call data memory from filling up too quickly.

Further help under Call data

Successful

Incoming calls that resulted in a connected call.

Further help under <u>Call data</u>

Only with charges

Calls that cause charges (e.g. collect calls).

Further help under <u>Call data</u>

Unsuccessful

Incoming calls that do not result in a connected call.

Further help under Call data

Charges independent from calls

Calls with additional charges (e.g. the transfer charge for collect calls).

Contacts and assignment

Contacts and assignment Contact Groups Authorisations Settings

Contacts and assignment

Contacts
Contact groups and Assignment
Google Contact Groups
CSV import group
Edit

Contacts



All contacts

All available contacts are displayed.

Searching for contacts can be restricted in the following way:

- Using the filter by entering at least one character or digit.
- By selecting a contact group under Contact groups and Assignment.

Further help under Contacts/LDAP

Filter

The list is filtered by the entered or selected value.

...

Opens a list of options for further view and editing of the contact.

Details

Displays details of this contact such as addresses and telephone numbers.

/ Edit

Opens the pages for editing the contact.

Details about settings under <u>Edit</u>

Contacts

- Note: Contacts in the CSV import group can not be edited! First adopt the CSV contacts into the contacts of the PBX.Adopting contacts from CSV import group.
- Note: Contacts from Google contact groups cannot be edited! Changes are only possible in the Gmail account.

QR code

Displays the QR code of this contact.

Exporting vCard v2.1

Exports a vCard in version 2.1.

Further help under Contacts/LDAP

+ New

Creates a new contact.

Details about settings under <u>Edit</u>

m Delete

Deletes the selected contacts.

- Important: Deleting a contact cannot be undone.
- Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

丛 Import

vCard v2.1 import

Imports a vCard in version 2.1.

CSV import

Imports contacts in CSV format, e.g. exported from another PBX or another contact database. The imported contacts can be transferred to the contacts of the PBX and then assigned to one or more contact group(s).

File name

Select a file in the csy format

Template

Select a suitable import template.

Preview

Display the entries in the CSV file and their assignment.

Further help under Contacts/LDAP

m Delete all

Only Admin: Deletes all existing contacts simultaneously.

Important: If you delete all the contacts you cannot undo this step.

↑ CSV export

Exports a contact group, individual or all contacts in a csv file.

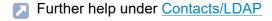
Important: Note the following legal notes.

It might be that the contacts include the following legally especially protected components:

- Private and work address
- · Private and work phone and fax numbers
- The date of birth

Contacts

- A photo of the person (in connection with the data backup)
- The electronic contact addresses (private and work), e.g. the email address
- The position and department within a company Before the contacts are being saved, all users of the PBX must be informed that these contents will be forwarded and about who will use the data received. It is essential to store the data securely and in accordance with the legal requirements. If you want to make the contacts available to your specialised dealer or to the manufacturer for error analysis, you must make sure that the legal requirements are fulfilled.



Contact groups and Assignment

Contact groups and Assignment

Display of contacts of the selected group. Contacts from other views can be assigned directly to this group by dragging them onto the graphic.

Searching for contacts can be restricted in the following way:

- · Using the filter by entering at least one character or digit.
- Further help under Contacts/LDAP

Filter

The list is filtered by the entered or selected value.

•••

Opens a list of options for further view and editing of the contact.

Details

Displays details of this contact such as addresses and telephone numbers.

/ Edit

Opens the pages for editing the contact.

- Details about settings under <u>Edit</u>
- Note: Contacts in the CSV import group can not be edited! First adopt the CSV contacts into the contacts of the PBX.Adopting contacts from CSV import group.

Note: Contacts from Google contact groups cannot be edited! Changes are only possible in the Gmail account.

QR code

Displays the QR code of this contact.

Exporting vCard v2.1

Exports a vCard in version 2.1.

Further help under Contacts/LDAP

m Delete contacts in contact group

Deletes the selected contacts from this or all contact groups. In addition, there is the option to permanently remove the selected contact.

- Important: Deleting a contact cannot be undone.
- Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

↑ CSV export

Exports a contact group, individual or all contacts in a csv file.

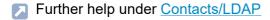
Important: Note the following legal notes.

It might be that the contacts include the following legally especially protected components:

- · Private and work address
- Private and work phone and fax numbers
- The date of birth
- A photo of the person (in connection with the data backup)

Contact groups and Assignment

- The electronic contact addresses (private and work), e.g. the email address
- The position and department within a company
 Before the contacts are being saved, all users of the PBX must be
 informed that these contents will be forwarded and about who will
 use the data received. It is essential to store the data securely and
 in accordance with the legal requirements. If you want to make the
 contacts available to your specialised dealer or to the manufacturer
 for error analysis, you must make sure that the legal requirements
 are fulfilled.



Google Contact Groups

BGoogle Contact Groups

Display of contacts of the selected google contact group. The contacts cannot be changed manually. Even after assignment to another contact group, the connection to the Gmail account remains intact and the contacts continue to be synchronised.

Searching for contacts can be restricted in the following way:

- · Using the filter by entering at least one character or digit.
- Further help under Contacts/LDAP

Q Filter

The list is filtered by the entered or selected value.

•••

Opens a list of options for further view and editing of the contact.

Details

Displays details of this contact such as addresses and telephone numbers.

/ Edit

Opens the pages for editing the contact.

Details about settings under <u>Edit</u>

- Note: Contacts in the CSV import group can not be edited! First adopt the CSV contacts into the contacts of the PBX.Adopting contacts from CSV import group.
- Note: Contacts from Google contact groups cannot be edited!
 Changes are only possible in the Gmail account.

QR code

Displays the QR code of this contact.

Exporting vCard v2.1

Exports a vCard in version 2.1.

Further help under Contacts/LDAP

m Delete contacts in contact group

Deletes the selected contacts from this or all contact groups. In addition, there is the option to permanently remove the selected contact.

- Important: Deleting a contact cannot be undone.
- Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

Exports a contact group, individual or all contacts in a csv file.

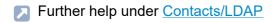
Important: Note the following legal notes.

It might be that the contacts include the following legally especially protected components:

· Private and work address

Google Contact Groups

- Private and work phone and fax numbers
- The date of birth
- A photo of the person (in connection with the data backup)
- The electronic contact addresses (private and work), e.g. the email address
- The position and department within a company
 Before the contacts are being saved, all users of the PBX must be
 informed that these contents will be forwarded and about who will
 use the data received. It is essential to store the data securely and
 in accordance with the legal requirements. If you want to make the
 contacts available to your specialised dealer or to the manufacturer
 for error analysis, you must make sure that the legal requirements
 are fulfilled.



CSV import group

CSV import

Displays all the imported CSV contacts that have been cached in the PBX system. The contacts can be added in **All contacts** view or directly into a contact group.

Searching for contacts can be restricted in the following way:

- Using the filter by entering at least one character or digit.
- Further help under Contacts/LDAP

Filter

The list is filtered by the entered or selected value.

•••

Opens a list of options for further view and editing of the contact.

Details

Displays details of this contact such as addresses and telephone numbers.

/ Edit

Opens the pages for editing the contact.

- Details about settings under <u>Edit</u>
- Note: Contacts in the CSV import group can not be edited! First adopt the CSV contacts into the contacts of the PBX.Adopting contacts from CSV import group.

Note: Contacts from Google contact groups cannot be edited!

Changes are only possible in the Gmail account.

QR code

Displays the QR code of this contact.

Exporting vCard v2.1

Exports a vCard in version 2.1.

Further help under Contacts/LDAP

in Delete contacts in the CSV contact group

Deletes the selected contacts from the CSV import group.

- Important: Deleting a contact cannot be undone.
- Note: You can select and delete multiple contacts at the same time by holding down the Shift or Ctrl key.

m Delete all from CSV import group

Deletes all contacts from the CSV import group.

Important: Deleting a contact cannot be undone.

CSV export

Exports a contact group, individual or all contacts in a csv file.

Important: Note the following legal notes.

It might be that the contacts include the following legally especially protected components:

CSV import group

- · Private and work address
- · Private and work phone and fax numbers
- The date of birth
- A photo of the person (in connection with the data backup)
- The electronic contact addresses (private and work), e.g. the email address
- The position and department within a company
 Before the contacts are being saved, all users of the PBX must be
 informed that these contents will be forwarded and about who will
 use the data received. It is essential to store the data securely and
 in accordance with the legal requirements. If you want to make the
 contacts available to your specialised dealer or to the manufacturer
 for error analysis, you must make sure that the legal requirements
 are fulfilled.
- Further help under Contacts/LDAP

Transfers individual, several or all contacts from the CSV import group to existing contact groups or in **All contacts**.

Edit

Contacts
E-mail addresses / Internet
Addresses
Contact Groups

Contacts

Personal details

A contact must at least consist of a phone number and a corresponding entry in the entry fields Surname, Display name or Company.

Additionally, you can add further phone numbers, e-mail and Internet addresses, postal addresses and a contact image. Furthermore, a contact is assigned to one or more contact group(s).

For the displaying in the call data list and for various phones, the **Display** name is used. If you have not entered anything in the entry field **Display name**, the display name will automatically be generated from the entries in the First name and Surname entry field. If no data is present neither in the First name nor Surname entry field, the display name is generated from the entry in the **Company** entry field.

Note the following differences when using the contact data in the connected phones:

COMfortel 1400 IP / 2600 IP / 3600 IP / 3200 / 3500

In the contact list of the phone almost all contact data will be displayed. Not displayed are **Display name**, **Department name** and **Date of birth**. Exception: If neither First name nor Surname has been entered, the Display name is displayed.

If there is already a contact entry with the same phone numbers in the local address book, the contacts will be merged in the phone. The local address book name entry is preserved.



Phone numbers

Up to four telephone numbers can be assigned and each one can be assigned to a category. Assign an individual name in the category **User-defined**.

Further help under Contacts/LDAP

+ New

Creates a new contact.

Further help under Contacts/LDAP

m Delete

Deletes this contact.

Important: Deleting a contact cannot be undone.

Edit image

Requirements:

· Saved contact

Creates an image to the contact.

, **Import**

Import of an existing graphic file. In order to obtain a good display quality the minimum size must be 96 x 96 pixel.

You can adjust the image section using the preview before saving.

m Delete image

Deletes the image set to this contact.

Important: Deleting the image cannot be undone.

E-mail addresses / Internet

E-mail addresses

Up to three e-mail addresses can be assigned and each one can be assigned to a category. Assign an individual name in the category **User-defined**.

Further help under Contacts/LDAP

Internet

Up to three Internet addresses can be assigned and each one can be assigned to a category. Assign an individual name in the category **User-defined**.

Further help under Contacts/LDAP

+ New

Creates a new contact.

Further help under Contacts/LDAP

m Delete

Deletes this contact.

Important: Deleting a contact cannot be undone.

Edit image

Requirements:

· Saved contact

Creates an image to the contact.

Import

Import of an existing graphic file. In order to obtain a good display quality the minimum size must be 96 x 96 pixel.

You can adjust the image section using the preview before saving.

Further help under Contacts/LDAP

m Delete image

Deletes the image set to this contact.

Important: Deleting the image cannot be undone.

Addresses

Addresses

Up to three addresses can be assigned and each one can be assigned to a category. Assign an individual name in the category **User-defined**.

Further help under Contacts/LDAP

+ New

Creates a new contact.

Further help under Contacts/LDAP

m Delete

Deletes this contact.

Important: Deleting a contact cannot be undone.

Edit image

Requirements:

Saved contact

Creates an image to the contact.

Import of an existing graphic file. In order to obtain a good display quality the minimum size must be 96 x 96 pixel.

You can adjust the image section using the preview before saving.

Addresses

Further help under Contacts/LDAP

m Delete image

Deletes the image set to this contact.

Important: Deleting the image cannot be undone.

Contact Groups

Contact Groups

Assigns this contact to one or more contact groups.

Further help under Contacts/LDAP

Define selected contact groups as standard

The selected contact groups are defined as standard.

Further help under Contacts/LDAP

+ New

Creates a new contact.

Further help under Contacts/LDAP

m Delete

Deletes this contact.

Important: Deleting a contact cannot be undone.

Edit image

Requirements:

Saved contact

Creates an image to the contact.



Contact Groups

Import of an existing graphic file. In order to obtain a good display quality the minimum size must be 96 x 96 pixel.

You can adjust the image section using the preview before saving.

Further help under Contacts/LDAP

m Delete image

Deletes the image set to this contact.

Important: Deleting the image cannot be undone.

Contact Groups

Overview
Configure Google contact group
Authenticate Google contact group

Overview

Contact Groups

Freely selectable name. This name is used to identify the group within the PBX.

These contact groups will only be transferred to the LDAP server and therefore to the IP end devices, if the system-wide automatic exchange line request is enabled.

Further help under Contacts/LDAP

Type

The type is selected during creation. A subsequent change is not possible.

Interface

Allows free assignment of contacts created manually, taken from the call data list or imported via vCard or CSV.

Google

Allows synchronisation with a Gmail account.



Further help under Contacts/LDAP

Contacts (Stored/)

Specifies how many contacts are existing in the contact group.



Contacts (/Maximum)

Specifies how many contacts could be existing in the contact group at maximum.

Further help under <u>Contacts/LDAP</u>

Password

Freely selectable password.

The user name resulting from the name of the contact group along with the associated password is needed for authentification of IP end devices on the LDAP server so that they have (only read) access to the contact data for this contact group.

- Note: To change the current password, click on ⊚ Edit passwords.
 After you have entered your PIN or password, click on ⊚ Hide passwords.
- Further help under Contacts/LDAP

Last synchronisation

Displays the time on which the Google contact group was last synchronised with the Gmail account's contact data, either automatically or manually.

Further help under Contacts/LDAP

•••

Opens various options for the Google contact group.

Configure

Opens the configuration of the Google contact group.

- Details about settings under <u>Configure Google contact group</u>
- Note: If Google authentication has not yet been carried out, call up the required dialogue here.
 - Details about settings under <u>Authenticate Google contact group</u>

Manual synchronisation

Starts the manual synchronisation of the Google contact group.

Further help under Contacts/LDAP

+ New

Creates a new contact group.

The type is selected during creation. A subsequent change is not possible.

Interface

Allows free assignment of contacts created manually, taken from the call data list or imported via vCard or CSV.

Google

Allows synchronisation with a Gmail account.

- Note: When the Google contact group is selected, Google authentication starts.
 - Details about settings under <u>Authenticate Google contact group</u>
- Further help under Contacts/LDAP



Deletes the entries marked in the list.

Configure Google contact group

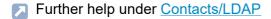
Google synchronisation active

The google contact group is synchronised with the contact information of the Gmail account at the set interval. Alternatively, the synchronisation can also be started manually.



Last synchronisation

Displays the time on which the Google contact group was last synchronised with the Gmail account's contact data, either automatically or manually.



Interval

Determines after what time the Google contact group is synchronised with the contact data of the Gmail account.

Further help under Contacts/LDAP

Contacts used

Determines which contact data of the Gmail account (individual group or all) are synchronised with the Google contact group.

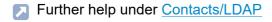
X Close

The dialogue for the configuration is closed, already created data is discarded.

Authenticate Google contact group

Release code -

Copies the release code provided by Google to the clipboard so it can be used for authentication.



Authentication

You are transferred directly to Google, so you can authenticate your Google account. Input the required data in the authentication dialogue.

Further help under Contacts/LDAP

× Close

The dialogue for the configuration is closed, already created data is discarded.

Authorisations

Contact Group

Select the contact group you want to display/configure.

Further help under Contacts/LDAP

Filter

The list is filtered by the entered or selected value.

ΑII

Activates all options simultaneously (Read access, Write access, Standard contact group).

Further help under Contacts/LDAP

Read access

Allows read access to this contact group. If read access is available, users can transfer the contacts contained in the contact group to their IP end devices via LDAP.

Read access also includes the following functions after logging into the web interface:

- Note: Login into the web interface is currently only available for the role of admin.
 - · search for and display a contact
 - read the QR code
 - export a contact data record (vCard)

Further help under Contacts/LDAP

Write access

Allows write access to this contact group.

Write access includes the following functions after logging into the web interface:

- Note: Login into the web interface is currently only available for the role of admin.
 - · create contacts
 - import contact images
 - · edit contacts
 - import contact data sets (vCard)
 - · adopt contacts from the call data list
 - · delete contacts
 - import contacts
 - export contacts
 - switch on/off the use as standard contact group
 - · assign contacts to a contact group
 - · delete contacts from a contact group
- Further help under Contacts/LDAP

Standard contact group

New contacts created by the admin/user are automatically transferred to this contact group.

Authorisations



Settings

LDAP server
Transfer via LDAP Server
Conversion

LDAP server

Server information

The PBX's LDAP server is an LDAP database with the contact data of the PBX's central contact files. As the LDAP server displays the contact data stored in the central contact files, it can also hold up to 2000 contacts like the files

The activation of the LDAP server allows to transfer contact data to the IP end devices.



Further help under Contacts/LDAP

Simplified user name

Freely selectable part of the user name. The user name along with the associated password is needed for authentification of IP end devices on the LDAP server so that they have (only read) access to all the contact data that is currently present on the LDAP server.



Further help under Contacts/LDAP

Password

Automatically generated, freely changeable password. Along with the user name, it is needed for authentification of IP end devices on the LDAP server so that they have (only read) access to all the contact data that is currently present on the LDAP server.

Note: To change the current password, click on

 ■ Edit passwords. After you have entered your PIN or password, click on • Hide passwords.

LDAP server

Transfer via LDAP Server

Information for configuring via IP end device

For the transfer of contact data from the LDAP server of the PBX to the IP end device, the values displayed in the web interface have to be entered into the end device.

All contact groups, selected contact groups or contacts of individual users can be transferred.



Conversion

Display name conversion

If you did not enter any data in the **Display name** field when you created a contact, the display name is generated automatically from the entries in the **First name** and **Surname** entry field.

Surname, First name

Display names are formed from the surname and the first name (separated by a comma and a blank space).

First name Surname

Display names are formed from the first name, a blank space and the surname.

- Note: If neither the First name nor Surnamehas been entered for the contact, the diplay name is generated from the Company entry field.
- Further help under Contacts/LDAP

Messages

Messages

Select the voice mail or fax box on the basis of their number/name.

- Further help under Voice mailbox
- Further help under Fax box

Filter

all

Shows all messages/faxes including those in the archive.

New

Shows only new incoming messages/faxes.

today up to last month

Shows all messages/faxes received in the specified time which have not yet been archived.

Archive

Shows all messages/faxes moved to the archive.

- Further help under Voice mailbox
- Further help under Fax box

Direction

all

Shows all messages/faxes including those in the archive.

received

Shows only received messages/faxes.

sent

Shows only sent messages/faxes.

- Further help under Voice mailbox
- Further help under Fax box

Search (phone number or name)

Phone number

Only messages/faxes are displayed which contain the searched telephone number in the following columns: **Caller**, **Call destination**, **Line**.

Phone number ranges

Only messages/faxes are displayed which contain the searched telephone number range (e.g. **00** for international calls) in the following columns:**Caller**, **Call destination**, **Line**.

Name

Only messages/faxes are displayed which contain the searched name in the following columns:**Name**, **Line**.

Name fragment

Only messages/faxes are displayed which contain text with the searched character string in the following columns:**Name**, **Line**.

Note: An option set under Filter will not be considered. All messages corresponding to the search term will be shown in any case.

Messages

- Note: During the fax transfer the status of the transfer when calling the pages is displayed under **Options**. It is not updated automatically. Use **Refresh** to reload the page and update the data displayed.
- Note: If the fax could not be transferred, a status message will be output.
- Further help under Voice mailbox
- Further help under Fax box

Status

Shows the status of the message:

- Unread
- Read
- Further help under Voice mailbox
- Further help under Fax box

Date / time

Point of time of receiving the message.

- Further help under Voice mailbox
- Further help under Fax box

Duration

Voice mailbox: duration of the recorded message in the format hh:mm:ss.

Further help under Voice mailbox

SD card

Fax box: number of pages of the received fax message.

Further help under <u>Fax box</u>

Caller

Voice mailbox: phone number of the caller for this recorded message.

Fax box: phone number of the sender for this incoming fax message.

- Further help under Voice mailbox
- Further help under Fax box

Name

Voice mailbox: name of the caller for this recorded message.

Fax box: name of the sender for this incoming fax message.

- Further help under Voice mailbox
- Further help under Fax box

Call destination

Telephone number of the receiving user.

- Further help under Voice mailbox
- Further help under Fax box

Line

Displays the own phone number.

Voice mailbox: phone number dialled by the caller.

Fax box: phone number that has been transmitted by the PBX.

- Further help under Voice mailbox
- Further help under Fax box

	Marks the message as unread.
8	Moves, copies or archives the messages/faxes.
⊥	Sends the message/fax via e-mail. If the configurations for e-mail transfer have not yet been completed (e.g. missing Message recipient, the message will not be sent via e-mail.
	Opens or saves the message/fax. Requirement for opening: an installed audio player for playing the voice message or an installed Acrobat Reader for reading the fax.
	A fax could not be sent. A status message will be output.

Further help under Voice mailbox

Messages

Further help under Fax box

C Refresh

The overview of the voice mail and fax boxes is refreshed.

- Further help under Voice mailbox
- Further help under Fax box

Entries marked in the list are set to read.

- Further help under Voice mailbox
- Further help under Fax box

Unread

Entries marked in the list are set to unread.

- Further help under Voice mailbox
- Further help under Fax box

Moves, copies or archives the messages/faxes.

- Further help under Voice mailbox
- Further help under Fax box

Administration

Contact information

Licences

Date and Time

Network

VoIP

Certificates

E-mail

Service

Maintenance

Contact information

Administrator access Interface
Operator and Location
Dealer

Administrator access

Administrator access | / Edit

Requirements:

· Knowledge of the current admin password

The user data for the administrator access are used for administrator registration on the PBX.

- Note: Changes at the administrator access are only possible with an encrypted HTTPS connection.
- Note: After a change, a new registration is required.
- Further help under <u>Access authorisations</u>

Admin user name

Freely selectable name. Is used for the Registration as admin on the web interface.

Further help under Access authorisations

Change admin PIN

Freely selectable PIN. Is used for operating some settings using the telephone.

Further help under Access authorisations

Change admin password

Freely selectable password. Is used for the Registration as admin on the web interface.



Further help under Access authorisations

Interface

Interface

Name

Freely selectable name. This name is used for identification of the PBX.

Description

Additional PBX information.

Operator and Location

Company... Note

User and location data of the PBX.

Dealer

Dealer

Company... homepage

Data of the specialised dealer or qualified electrician who commissioned the PBX and is servicing it.

Licences

Licences

Available Licences
Initialising the system again
Activation code for extended functions

Available Licences

Function

Overview of the enabled functions and the feature level.

- Further help under <u>Licences</u>
- Further help under Feature level

C Refresh

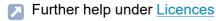
The overview of the available licenses is refreshed.

Further help under <u>Licences</u>

Initialising the system again

Initialisation code for the license server

Generating the initialisation code.



Activation code for extended functions

Purchase voucher code 🕹

Opens the online shop.

Further help under <u>Licences</u>

Redeem voucher manually \$\frac{1}{2}

Opens the Voucher Centre.

- **Important:** When you change a voucher code into an activation code, the right of withdrawal expires before expiry of the withdrawal period.
- Further help under Licences

Update activation license 🗏

Starts an adjustment of the licences outside of the regular adjustment. This is e.g. useful after purchasing additional licences.

Further help under Licences

Entering activation license

Opens a field for entering the activation licence. This is only necessary if the licences cannot be compared.

Further help under <u>Licences</u>

Date and Time

Date and Time
One-time configuration
Automatic synchronisation
Automatic distribution

Date and Time

Current date / current time

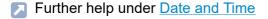
The current display of the date and time set in the PBX. Date and time depend on the setting under **Time zone**.

Time zone

The date and time of the PBX are set depending on the time zone selected here. The time zone should therefore be chosen to match the location of the facility.

UTC

Coordinated world time, without deviation.



One-time configuration

Manually | **⊘** Set now

Takes the date and time from the values entered here.

Time

Enter in format: hh:mm:ss

Date

Enter in format: dd.mm.yyyy

Further help under Date and Time

Take from PC | ⊗ Set now

Takes the date and time from the currently valid PC time.

Further help under Date and Time

Automatic synchronisation

From the Internet from an NTP server

The date and time are queried and synchronised by the NTP servers set here. The NTP pool set as default depending on the location is usually sufficient for correct time determination.

- Note: To prevent time determination via NTP server (not recommended), the entries must be deleted here. Any updates (e.g. time changeover, adjustment after power failure or system reset) must then be actively initiated by the system administrator.
- Further help under Date and Time

Automatic distribution

NTP server service

The IP address of the PBX is provisioned as an NTP server in the associated telephones/devices. If the devices are in the same network as the PBX, they subsequently obtain their time from the PBX.

- **Note:** If the service is switched off, another available address is provisioned into the devices as the NTP server instead.
 - NTP server 1, if set up. An address that can be reached by the PBX and its devices should therefore be entered here.
 - Alternatively NTP pool according to the location of the PBX
- Important: If the units are neither in the network of the PBX nor are they able to reach addresses on the Internet, an individually created provisioning file is required in which an accessible NTP server is entered.
- Further help under <u>Date and Time</u>

Network

IP configuration
Server service
Gateways and routing
VPN
IP block- and allowlist
Ports

IP configuration
NET module
DHCP server settings for NET-module
DNS configuration
HTTP proxy configuration

Obtain IP address automatically (DHCP client)

The PBX automatically receives its IP address from the DNS server.

Further help under Network

Assigning IP address manually

Requirements:

Disabled DHCP client

IP address

IPv4 address in the range 0 - 255 (example: 192.168.0.240).

Subnet mask

IPv4 address of the subnet mask in the range 0 - 255 (example: 255.255.255.0).

Gateway

IPv4 address of the gateway in the range 0 - 255 (example: 192.168.0.2).

Further help under Network

Permanent IP address from the APIPA address range

The PBX can be reached via the IP address 169.254.1.240, if

- it is directly connected to an individual computer which automatically receives its IP address.
- it is connected to a router that automatically assigns IP addresses.

Further help under Network

MAC address

Displays the specific and unique hardware address of the PBX.

Further help under Network

IPv6 configuration | /

Opens the IPv6 configuration.

IPv6 Mode

If IPv6 is switched on, an IP address complying with version 6 of the Internet protocol can be assigned to the PBX. Thanks to the greater length of IPv6 addresses (eight blocks with four digits each, example: 3001:00FF:ABC0:0EAC:0001:0000:0000:000F), IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

deactivated

Switches off the use of IPv6 addresses.

SLAAC

Automatically assigns an IPv6 address to the PBX.

static

Assigns a user-defined IPv6 address to the PBX.

Note: If IPv6 has been disabled for the PBX, IPv6 can also not be enabled for any modules. If, in the reverse, IPv6 has been enabled for

the PBX, it is also enabled automatically for all modules and cannot be disabled.

Further help under Network

IPv6 address

Requirements:

IPv6 modestatic selected

IPv6 address

IPv6 address of the PBX.

Further help under Network

Prefix length

Requirements:

IPv6 modestatic selected

Prefix length

Prefix length in the range **1** up to **128**. The prefix specifies to which subnet the IPv6 address belongs.

Further help under Network

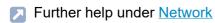
IPv6 gateway

Requirements:

IPv6 modestatic selected

IPv6 gateway

Possible entries: Digits, characters and :.



NET module

Obtain IP address automatically (DHCP client)

The NET module obtains its IP address automatically from a DHCP server.

- Further help under Network interface (second)
- Further help under Network

Assigning IP address manually

Requirements:

Disabled DHCP client

IP address

IPv4 address in the range 0 - 255 (example: 192.168.0.240).

Subnet mask

IPv4 address of the subnet mask in the range 0 - 255 (example: 255.255.255.0).

- Further help under Network interface (second)
- Further help under Network

MAC address

Specific and unique hardware address of the NET module.

- Further help under <u>Network interface (second)</u>
- Further help under Network

IPv6 configuration | /

Opens the IPv6 configuration.

IPv6 Mode

If IPv6 is switched on, an IP address complying with version 6 of the Internet protocol can be assigned to the NET module. Thanks to the greater length of IPv6 addresses (eight blocks with four digits each, example: 3001:00FF:ABC0:0EAC:0001:0000:0000:000F), IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

deactivated

Switches off the use of IPv6 addresses.

SLAAC

Automatically assigns an IPv6 address to the NET module.

static

Automatically assigns a user-defined IPv6 address to the NET module.

- Note: If IPv6 has been disabled for the PBX, IPv6 can also not be enabled for any modules. If, in the reverse, IPv6 has been enabled for the PBX, it is also enabled automatically for all modules and cannot be disabled.
- Further help under <u>Network interface (second)</u>
- Further help under Network

IPv6 address

Requirements:

IPv6 modestatic selected

IPv6 address

IPv6 address of the NET module.

- Further help under Network interface (second)
- Further help under Network

Prefix length

Requirements:

IPv6 modestatic selected

Prefix length

Prefix length in the range **1** up to **128**. The prefix specifies to which subnet the IPv6 address belongs.

- Further help under Network interface (second)
- Further help under Network

IPv6 gateway

Requirements:

IPv6 modestatic selected

IPv6 gateway

Possible entries: Digits, characters and :.

NET module

- Further help under Network interface (second)
- Further help under Network

DHCP server settings for NET-module

Use DHCP server for NET module

Requirements:

 If internal and external VoIP telephony are to be connected at the same network interface: Disabled Obtain IP address automatically (DHCP client) (page Administration > Network > IP Configuration under NET module)

When connecting SIP devices to the 2nd network interface, it is advisable to enable the DHCP server in order to supply the SIP devices with IP addresses and other information (NTP server address, gateway address etc.).

- Further help under Network interface (second)
- Further help under Network

Begin with DHCP address range

Begin of the IP address range from which the IP addresses are to be assigned.

Only IP addresses within the changeable range of the corresponding network class can be entered.

- Further help under Network interface (second)
- Further help under Network

End of DHCP address range

End of the IP address range from which the IP addresses are to be assigned.

Only IP addresses within the changeable range of the corresponding network class can be entered.

- Further help under <u>Network interface (second)</u>
- Further help under Network

DHCP configuration

Opens the **DHCP config editor** for importing and editing the configuration.

Import

A configuration saved on the data storage device (PC) can be selected and is saved onto the PBX. The configuration can be edited in the **DHCP config editor**.

- Further help under Network interface (second)
- Further help under Network

Default

All settings made will be reset to the standard values.

DNS configuration

Use DNS for NET module

Further help under Network

Use own DNS server

Requirements:

DHCP enabled

The here configured DNS server is used. A DNS server provided by the DHCP server is ignored.

- Note: If DHCP is disabled, the here configured DNS server is always used.
- Further help under Network

First DNS server / Second DNS server

Requirements:

• DHCP disabled or DHCP enabled and Use own DNS serveractivated

Address of the DNS server. The provider or network system administrator sets the DNS server. The second address is available.

IPv4 addressFour blocks separated by a dot (.).

IPv6 addressEight blocks separated by a colon (:).

Further help under Network

Test connection of the 1st network interface card to www.google.de |



Checks the connection via the set DNS server. With the command Ping, the PBX checks whether www.google.de can be reached. www.google.de has been chosen because it can usually be reached.



Further help under Network

HTTP proxy configuration

HTTP proxy

The connection to the Internet with a HTTP protocol via a proxy server is established.

Further help under Network

IP address or URL

Address of the proxy server.

Further help under Network

Port

Port of the proxy server.

- Note: If you do not enter anything in this entry field, port 3128 will be entered automatically.
- Further help under Network

Access data

Authorisation on the proxy server is necessary.

Further help under Network

User name

User name for the authorisation on the proxy server.

HTTP proxy configuration

Further help under Network

Password

Password for the authorisation on the proxy server.

- Further help under <u>Access authorisations</u>
- Further help under Network

Server service

Web server configuration Provisioning LAN TAPI SFTP server

Web server configuration

HTTP-Port

Port 80 is used for accessing the Web server by default. The address can thus be entered as follows: http://192.168.55.13, for example. Only when accessing via Ethernet the value (e.g. 8081 instead of 80) has to be entered when modifying the port (e.g. http://192.168.55.13:8081).

You can view the ports already configured under **Administration** > Network > Ports.



Caution: For security reasons, a port change is particularly required if the PBX is meant to be accessed over the Internet (over a router and a permanent IP address).

HTTPS port

Port 443 is used for accessing the web server by default. The address can thus be entered as follows: http://192.168.55.13, for example. Only when accessing via Ethernet the value (e.g. 4432 instead of 443) has to be entered when modifying the port (e.g. https://192.168.55.13:8081).

You can view the ports already configured under **Administration** > Network > Ports



A Caution: For security reasons, a port change is particularly required if the PBX is meant to be accessed over the Internet (over a router and a permanent IP address).

Certificate

All properties of the certificate are displayed. The certificate can be exported and saved to the (PC) data storage device.

Force HTTPS connection

The connection is carried out via HTTPS to secure the access to the web interface. An access via HTTP will be forwarded to HTTPS.

Note: When switched off, the connection is carried out via HTTP.

Access only within local network

For increased security, the web server access is limited to the local network.

API access only within local network

For increased security, the API access is limited to the local network.

Forced logout if no action happens after

Specifies after how many minutes a user is automatically logged out of the web interface if no action has been taken in the web interface within the configured time period.

Provisioning

Provisioning server active

Certain IP phones and DECT base stations are automatically configured via the PBX. The PBX transmits as provisioning server the configuration data to the device.

Requirement for the transfer is that the device has been created in the PBX and that a provisioning file has been assigned.

Further help under Provisioning

Auto provisioning active

If an IP telephone or DECT base station is connected with the PBX, the PBX checks on request if there is a suitable template for this device type. In the case of a match, the PBX transfers the URL of the provisioning server to the device and assigns the suitable provisioning file to the device. Configuration data is transferred to the device.

- Note: If there is another PBX with connected devices in the same network, it is recommended to keep this option disabled. Autoprovisioning can only be switched on after all devices in the network have been checked for the necessity of automatic configuration (and the associated implementation of appropriate measures).
- Further help under Provisioning

Access without client certificate (unsafe operation)

Only enable this function if you operate devices on the PBX that do not have an own client certificate.

Provisioning

- Further help under <u>Certificates</u>
- Further help under **Provisioning**

Provisioning only within local network

Provisioning is only carried out within the local network.

- Note: When switched off, provisioning can also be carried out on other possible networks (unsafe operation).
- Further help under Provisioning

Phone language after provisioning

Language that is displayed on the device after provisioning.

Further help under **Provisioning**

LAN TAPI

Network port of the TAPI server

Port on the PBX which is used for communications with the TAPI server.

- Note: The configuration does not take effect until you restart the PBX or the TAPI server.
- Further help under LAN TAPI

Password for the network connection of the TAPI server

Freely selectable password. Is used for communications with the TAPI server.

- Note: To change current passwords, click on Edit passwords. After you have entered your PIN or password, click on ○ Hide passwords.
- Further help under Access authorisations
- Further help under LAN TAPI

Restart TAPI server | C

Restarts the TAPI server.

Further help under LAN TAPI

Use PBX Call Assist

PBX Call Assist CTI software is responsible for the interaction between computer and telephony.

LAN TAPI

Further help under LAN TAPI

SFTP server

SFTP server

An external programme (e.g. evaluation programmes) can demand and download data (e.g. call data) using the SFTP server of the PBX. The SFTP network protocol is used for the secure data transfer between the PBX and external programs.

Further help under Call data

Password

Automatically generated, freely changeable password. Required by external programmes for registering at the SFTP server of the PBX an external programme. The **User name netstorage** predefined.

- Note: To change current passwords, click on Edit passwords.
 After you have entered your PIN or password, click on Hide passwords.
- Further help under Access authorisations

Gateways and routing

Internet Gateways IP routing

Internet Gateways

Name

Freely selectable name.

Further help under Network

IP address

Gateway address of the IAD.

Further help under Network

Interface

Requirements:

· LAN 2: Existing NET module

Select the network interface.

- Note: A default IAD with the data of the first network interface is automatically created.
- Further help under Network

+ New

Creates a new entry.

Further help under Network

IP routing

IP routing

IP routing allows selecting an alternate Internet gateway for sending IP packages, such as the connection to external private branch exchanges.

- Further help under Network interface (second)
- Further help under Network

Network address

Network or host address for the routing entry.

- Further help under Network interface (second)
- Further help under Network

Prefix length

Address range to be used for the routing entry.

- Further help under <u>Network interface (second)</u>
- Further help under Network

Gateway

Defines one of the previous set Internet gateways for the routing entry.

- Further help under Network interface (second)
- Further help under Network



Creates a new entry.

VPN

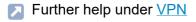
OpenVPN Server

The OpenVPN server configured here can be used to securely connect devices that support VPN to the PBX.



OpenVPN Configuration /

Opens the saved OpenVPN configuration in a code editor. The OpenVPN configuration as delivered must be supplemented and adapted.



IP block- and allowlist

IP blocklist IP allowlist

IP blocklist

IP address

The IP blocklist shows the IP addresses that are classified as dangerous and are therefore blocked. The blocked IP address is displayed with the blocking time and the reason reported by Fail2ban. Existing entries can be deleted or transferred to the IP allowlist.

Further help under Network

•••

m Delete

Deletes the entry from the IP blocklist.

Transfer

Transfers the entry to the IP allowlist.

Further help under Network

IP allowlist

IP address

The IP allowlist shows the IP addresses that are not classified as dangerous and are therefore not blocked. The list is freely configurable. Enter here, for example, the IP address of the computer with which you normally configure the PBX.

Further help under Network

+ Adding

Adds a single IP address or IP address range to the IP allowlist.

IP address ranges are entered using CIDR notation.

Example enter: 192.168.21.0/24

Subnet mask: 255.255.255.0

• Host ip (from): 192.168.21.0

• **up to**: 192.168.21.255

Further help under Network

m Delete

Deletes the entry from the IP allowlist.

Ports

Overview

Displays all configured and fixed ports of the PBX and the used transport protocols.



Further help under Network

VoIP

VoIP

VoIP configuration SIPS / SRTP internal

VoIP configuration

Presetting of the VoIP channels
Distribution of the VoIP channels
Proxy server for COMfortel SoftPhone
SIP
RTP
Quality of Service (QOS)
STUN server

Presetting of the VoIP channels

VoIP operating mode

Classic

All supported codecs are used. This heavily limits the number of available VoIP channels.

Broadband (only COMtrexx Business)

Priority is given the use of broadband codecs for the best possible VoIP call quality.

VolP oriented

Priority is given to a higher number of available VoIP channels. For this purpose, the amount of codecs is limited. Provides minimal support for classic communication technologies, for example for integrating fax machines or door terminals.

VoIP maximum

The operating mode with the highest possible number of available VoIP channels. For this purpose, the amount of codecs is heavily limited. No classic telephony is possible (analogue).

- Note: The actual avalaible VoIP channels are displayed after having selected VoIP operating mode.
- Further help under VoIP

Distribution of the VoIP channels

Channel reserved for voice mail/fax

Number of channels to be used for voice mail/fax and not for VoIP.

Further help under VolP

External VolP channels

Number of VoIP channels to be used only as external VoIP channels.

Further help under VolP

Internal VoIP channels

Number of VoIP channels to be used only as internal VoIP channels.

Further help under VolP

Available VoIP channels for dynamic assignment

Number of VoIP channels to be used as internal or external VoIP channels depending on the need.

Further help under VolP

Proxy server for COMfortel SoftPhone

Use proxy

Activates the proxy service for connecting the COMfortel SoftPhone.

- Note: Disable the proxy service if you are not using a COMfortel SoftPhone.
- Further help under **Softphone**

SIP proxy

Address of the Auerproxy server. In the default settings: proxy.auerproxy.de

- Note: Make changes only when requested to do so by a service representative.
- Further help under Softphone

Registration address

Registration address in the PBX for communication with the Auerproxy server.

Further help under Softphone

Local port

Local port used internally (default value: 5099).

Note: For encrypted connections to the Auerproxy server, the port with the value set here + 1 is used (default value: 5100).

Proxy server for COMfortel SoftPhone



Further help under Softphone

SIP

SIP port of the internal VoIP registrar

You will need to change this if one of the VoIP providers you use uses the default value 5060 set here.

- Important: If the SIP port is modified while it is running, all the calls will be interrupted.
- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Note: Change the SIP port in all VoIP phones that are connected to the PBX.
- Further help under VolP

SIPS port of the internal VoIP registrar

The internal SIPS port is calculated automatically from the SIP port + 1.

Further help under VolP

VoIP registrar Ethernet interface

Defines the network interface for the VoIP registrar and therefore, two possible usage scenarios:

LAN₁

Only one network interface

or

VoIP provider at the LAN 2 connection

LAN 2

Internal VoIP on LAN connection 2

or

VoIP completely on LAN connection 2

- Further help under Network interface (second)
- Further help under VolP

RTP

RTP Range

The RTP port is a port on the local system that is used as the outgoing port for RTP transfers.

- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Further help under VolP

Quality of Service (QOS)

DiffServ

Requirements:

Existing and enabled DiffServ support on all active network components

DiffServ is used to prioritise the handling of language packages (VoIP) and to achieve better voice quality for VoIP calls. DiffServ can be evaluated in networks that include active network components such as routers or switches, to enable packets to be forwarded according to their priority.

As proposed in RFC 4594, the DSCP values are set for VoIP calls (signalling (SIP): CS5, voice (RTP): EF).

- Note: DiffServ is usually not supported by VoIP providers. As a result, this setting may not be taken into consideration under certain circumstances on the Internet.
- Note: To adopt the change, reboot the PBX.
- Further help under VolP

STUN server

STUN server

Requirements:

- · support for public DNS mapping by the router on the PBX network
- call forwarding from the SIP port of the internal VoIP registrar to the PBX's IP address in the router
- Configured DNS server
- Note: If IP protocol IPv6 is selected, STUN cannot be used as a NAT method for neither SIP nor RTP.
- Further help under VolP

IP address or URL

Address of the STUN server

Further help under VolP

Port

Port of the STUN server.

- Note: The page Administration > Network > Ports shows an overview of the PBX ports (incoming and outgoing).
- Further help under VolP

SIPS / SRTP internal

SIPS / SRTP Root certificate PBX certificate

SIPS / SRTP

SIPS / SRTP

Internal VoIP call connections are encrypted to prevent listening in. The connection setup and termination, and also call signalling, are encrypted with SIPS. The call data is encrypted with SRTP.

The following matching keys and certificates are required for SIPS/SRTP internal:

- · valid root certificate
- · valid PBX certificate
- · private key of the PBX certificate
- Note: In the end devices used (SIP phones, SIP soft clients, DECT base stations), the encryption setting must match the setting in the PBX. If this is not possible in individual cases, an exception can be set here for the end device: Telephones > Assignment > / > RTP encryption
- Important: To change a certificate, SIPS/SRTP must be switched off.
 When switching it on again, it may happen that calls are interrupted.
- Further help under <u>Certificates</u>
- Further help under VolP

Root certificate

■Root certificate

Opens information about the root certificate.

Further help under Certificates

m Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

Further help under <u>Certificates</u>

Select a certificate that is stored on a data storage device and store it in the PBX.

Further help under <u>Certificates</u>

____ Export

The certificate stored in the PBX is downloaded from the device and saved to a (PC) data storage device.

Further help under Certificates

C Renew

The certificate is renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Root certificate

Further help under Certificates

Fingerprint

Fingerprint of the root certificate. The fingerprint clearly identifies the root certificate with a numerical code.

- **Note:** If the root certificate is renewed, a new fingerprint is generated automatically.
- Further help under Certificates

PBX certificate

■ PBX certificate

Opens information about the PBX certificate.

Further help under Certificates

m Delete

Deletes a certificate currently saved in the PBX after user confirmation at the prompt.

Further help under Certificates

Select a certificate that is stored on a data storage device and store it in the PBX.

Further help under <u>Certificates</u>

____ Export

The certificate stored in the PBX is downloaded from the device and saved to a (PC) data storage device.

Further help under Certificates

C Renew

The certificate is renewed. This affects, among other things, the period of validity and the fingerprint of the certificate.

Further help under Certificates

Private key

Private key is available

A private key is available in the PBX certificate.

No private key available

A private key is not available in the PBX certificate.

Note: If the PBX certificate is renewed, a new private key is generated automatically.

丛 Import

Select a certificate that is stored on a data storage device and store it in the PBX. An existing certificate will be overwritten.

Delete

Deletes the private key currently saved in the PBX after user confirmation at the prompt.

Further help under <u>Certificates</u>

Certificates

Not trustworthy
Trustworthy
Certificate authorities
Web server
SIPS / SRTP internal

Not trustworthy

Not trustworthy

List of certificates considered as untrustworthy. If checking of the server certificate against the root certificate (or stack of certificates) stored in the PBX fails, the connection will be interrupted and the certificate will be regarded as not trustworthy.

Further help under Certificates

Issuer

The column displays the issuer of the certificate.

Further help under Certificates

Issued on

The column displays when the certificate has been issued.

Further help under Certificates

Valid until

The column displays the validity period of the certificate.

Further help under <u>Certificates</u>

View

All properties of the certificate are displayed. The certificate can be exported and saved to the (PC) data storage device.

Not trustworthy

Further help under Certificates

■ Trust certificate

The marked certificate is considered trustworthy.

Further help under Certificates

Trustworthy

Trustworthy

List of certificates considered as trustworthy. If checking of the server certificate against the root certificate (or stack of certificates) stored in the PBX does not fail, the connection will be set up and the certificate will be regarded as trustworthy.

Further help under Certificates

Issuer

The column displays the issuer of the certificate.

Further help under <u>Certificates</u>

Issued on

The column displays when the certificate has been issued.

Further help under Certificates

Valid until

The column displays the validity period of the certificate.

Further help under <u>Certificates</u>

View

All properties of the certificate are displayed. The certificate can be exported and saved to the (PC) data storage device.

Trustworthy

Further help under Certificates

Import

Select a certificate that is stored on a data storage device and store it in the PBX.

Further help under Certificates

Certificate authorities

Certificate authorities

List of certificates of different certificate authorities known by the PBX.

Further help under Certificates

Issuer

The column displays the issuer of the certificate.

Further help under Certificates

Issued on

The column displays when the certificate has been issued.

Further help under <u>Certificates</u>

Valid until

The column displays the validity period of the certificate.

Further help under <u>Certificates</u>

View

All properties of the certificate are displayed. The certificate can be exported and saved to the (PC) data storage device.

Further help under Certificates

+ New

Individual certificates are added again or all certificates are restored.

Further help under Certificates

m Delete

Deletes the entries marked in the list.

Web server

Issuer

The column displays the issuer of the certificate.

Further help under <u>Certificates</u>

Issued on

The column displays when the certificate has been issued.

Further help under <u>Certificates</u>

Valid until

The column displays the validity period of the certificate.

Further help under Certificates

View

All properties of the certificate are displayed. The certificate can be exported and saved to the (PC) data storage device.

Further help under <u>Certificates</u>

SIPS / SRTP internal

Issuer

The column displays the issuer of the certificate.

Further help under <u>Certificates</u>

Issued on

The column displays when the certificate has been issued.

Further help under <u>Certificates</u>

Valid until

The column displays the validity period of the certificate.

Further help under Certificates

View

All properties of the certificate are displayed. The certificate can be changed.

- Details about settings under SIPS / SRTP internal
- Further help under <u>Certificates</u>

Delete

Deletes the entries marked in the list.

E-mail

E-mail

Outgoing mail server Logo management Internationalisation

Outgoing mail server

Outgoing Mail Server (SMTP)

Select the outgoing mail server to be displayed/configured.

Further help under E-mail

Default outgoing mail server

The selected outgoing mail server is used as default.

- Note: The outgoing mail server first configured in the PBX is automatically switched on. To switch on another outgoing mail server, the currently switched on outgoing mail server must first be switched off.
- Further help under E-mail

Name

Freely selectable name. This name is used to identify the outgoing mail server within the PBX and from connected phones.

Further help under E-mail

Address

Enter the address of the outgoing mail server:

IPv4 address

Four blocks separated by a dot (.).

IPv6 address

Eight blocks separated by a colon (:).

Further help under E-mail

Port (1..65535)

Requirements:

· Knowledge of the access data for the outgoing mail server

Enter the port number of the outgoing mail server between **1** up to **65535**. You can obtain the port number from your e-mail provider or read it up in the Internet.

Further help under E-mail

Encryption (SSL)

If encryption is enabled, the connection of the outgoing mail server is secured using the SSL (Secure Socket Layer) protocol. The data is transferred via https.

No encryption

Switches encryption off.

Encryption (SSL) after STARTTLS

Switches encryption on after STARTTLS.

Encryption (SSL) immediately

Switches encryption immediately on.

Further help under E-mail

Type of authentication

No authentication

Disables authentication.

User name / Password (AUTH PLAIN)

Enables authentication via user name/password.

User name / Password (AUTH LOGIN)

Enables authentication via user name/password.

MD5 challenge-response (AUTH CRAM-MD5)

Enables authentication via MD5 challenge-response (MD5 = Message-Digest Algorithm 5).

automatically

The authentication type is set automatically.

Further help under E-mail

User name

Requirements:

Enabled authentication

Enter the user name. In the most cases, this is your e-mail address.

Further help under E-mail

Password

Requirements:

Outgoing mail server

· Enabled authentication

Enter the password that belongs to the user name.

- Important: Space and the following special characters are currently not supported: ! @ # \$ % ^ &
- Further help under E-mail

Sender

Enter the address of the e-mail sender. It is used to display it to the recipient of the e-mails.

Further help under E-mail

+ New

Creates a new entry.

Further help under E-mail

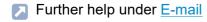
m Delete

Deletes the entries marked in the list.

Logo management

Memory utilisation

Displays the available storage capacity for logo files.



Existing logo files

Displays the list of imported logos.

Further help under E-mail

Select and import an image file (*.jpg) that is stored on a data storage device (PC).

The logo must be provided in *.jpg file format and must be 160 x 60 pixels (width x height) in size.

The maximum permitted file size for a logo is 50 kB. The memory for all logos is limited to 500 kB.

The name of the file must be unambiguous. It is displayed in the web interface and cannot be changed.

- Note: If you do not import your own logo, the Auerswald/FONtevo logo included in the scope of supply will be used.
- Further help under E-mail



Deletes the entries marked in the list.

Internationalisation

Overview Configure

Overview

Language file

The available language files (default language files, user-defined language files) are displayed.

Further help under E-mail

View

Opens the page to view the different text elements.

Configure

Opens the configuration of the user-defined language file.

- Details about settings under Configure
- Further help under <u>E-mail</u>

+ New

Creates a new entry.

Further help under E-mail

Name

Freely selectable name. This name is used to identify the language file within the PBX.

Further help under E-mail

Delete

Deletes the entries marked in the list.

Select and import a language file (*.xml) that is stored on a data storage device (PC).

The name of the file must be unambiguous. It is displayed in the web interface and cannot be changed.

The file structure of the imported language file must be identical to the file structure of the default language file. To this end, export a language file to your PC and Edit the text fields in the language template without changing the structure of the *.xml file.

Further help under <u>E-mail</u>

∴ Export

The list of providers is saved to the data storage device (PC) in an *.xml file.

Further help under E-mail

Configure

Language fileLanguage- Mail type

Select a text element category:

Scheduled call test e-mail

Text modules to generate a test e-mail.

Mailbox: New message

Text modules to generate an e-mail with information about a new message.

Mailbox: Automatic clean-up failed

Text modules to generate an e-mail regarding failure of automatic cleanup.

Mailbox: Storage space warning

Text modules to generate an e-mail when a specified storage capacity has been exceeded.

Mailbox: Fax transmission successful

Text modules to generate an e-mail regarding successful fax transmission.

Mailbox: Fax transmission failed

Text modules to generate an e-mail regarding failure of fax transmission.

System log

Text modules to generate an e-mail with system information, malfunctions and safety warnings.

General

General text elements

Configure

- Date:

Day:DD

Month: MM

Year: YYYY

- Time:

Hour (24-hour format):hh24

Hour (12-hour format): hh12

Minute:mm

Second:ss

am/pm (English language template only):ampm

Date and Time

Text modules with variables for Time/Date.

Further help under E-mail

Current content

Standard content of text elements in the language of the web interface.

Further help under E-mail

Translation

Own translation.

Further help under E-mail

Service

External access
Network Diagnosis
Logging

External access

Remote programming Access for CCC

Remote programming

Remote switching numbers

Displays the phone numbers set as remote switching numbers. Using these phone numbers, the PBX can be called from an external phone in order to operate a function using a key number, e.g. Follow-me.

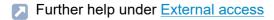


Further help under External access

Access for CCC

Confirm CCC access

Starts the requested pairing with the CCC. Subsequently, the PBX can be accessed via the CCC using the associated account in order to perform various service functions.



Monitoring

Allows monitoring via the CCC and thus access to various monitoring functions of the PBX (e.g. CPU utilisation, memory utilisation, etc.).

Further help under External access

Network Diagnosis

Trace Ping

Trace

> Start Trace

For controlling the network data stream, a network protocol in PCAP format can be recorded and is afterwards used for identifying disruptions and their causes (e.g. with Wireshark/Ethereal).

Further help under Service data

Ping

Target

Removed host (e.g. computer, PBX) in the network, its reachability and the routing time is to be checked.

Further help under Service data

Protocol

Protocol used for ping

Further help under Service data

Port

Port used for ping

Further help under Service data

Send command | Start

Sends the configured ping.

Further help under Service data

Console output

Response to the ping command.

Further help under Service data

Logging

Logging activated

A protocol can be recorded to log and display control and management information about the connections to the PBX e.g. to display errors. You will therefore be able to see the type of error, and maybe even its cause, from the list of transferred information.

- Important: If logging in PCAP format has been selected, anyone within the network can record a log over IP. This protocol also displays sensitive data, such as call connection data. Therefore, logging should only be activated if it is needed for error analysis purposes.
- Further help under Service data

Logging system

System for executing the logging. **Status:** Displays whether recording is taking place and from which address.

PCAP format (wireshark)

Logging in PCAP format. The recording can be done with a PC program that supports the PCAP format (e.g. Wireshark).

Syslog server

For this purpose configured syslog server. Logging of the connection data will start as soon as you have saved your settings.

Further help under <u>Service data</u>

Syslog server IP

IPv4 or IPv6 address of the syslog server.

Logging

Further help under Service data

Syslog server port

UDP port of the syslog server.

Further help under Service data

丛 Import

Import of a file in the supported formats *.cfc and *.cfg for activating certain log categories.

- Note: These log categories remain active only for the time configured in this file.
- Further help under Service data

Maintenance

Backup Update and Restart Message Configuration System messages Backup

Backup

Configuration
Remote Backup
Restoration points
Service data

Configuration

Configuration file | Import 🕹

Select and import a configuration file (*.cfg) that is stored on a (PC) data storage device. Some data packages are optional and can be selected for import.

- Important: Saving an old configuration overwrites the current configuration data.
- Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).
- Further help under Backup

Configuration file | Export 1

The configuration is saved to the (PC) data storage device as configuration file (*.cfg). Some data packages are optional and can be selected for export.

Further help under Backup

Configuration data

Configuration data will also be saved/restored in any case. Other components can be switched off so that the configuration file can also be saved in other PBXs.

Further help under Backup

Network settings

The network settings will also be saved/restored.

The network settings include the following data:

- IP configuration (DHCP client setting, IP address, Sub-network mask, Gateway)
- DNS configuration (First DNS server, Second DNS server)
- HTTP proxy configuration (HTTP proxy settings)
- · Port of the web server
- Further help under Backup

PBX data

The PBX data will also be saved/restored.

The PBX data include the following data:

- PBX name
- · Admin user name
- Contact information
- Admin PIN
- Admin password
- Further help under <u>Backup</u>

Music on Hold/announcements

The audio data available for music on hold and announcements will also be saved/restored

Configuration

Further help under Backup

Contact images

The contact images available for contacts will also be saved/restored.

Further help under Backup

Delete current configuration and start with a new configuration \mid reset \times

When resetting the configuration, the following parameters are maintained:

- · Admin user name, admin PIN and admin password
- IP address settings
- Web server port setting
- Time
- NTP server settings
- · Contact information
- Saved hold music and announcements
- · Call data
- · System messages
- · Restoration points

The following server services remain enabled:

- SLAPD
- NTPD

Configuration

Further help under **Backup**

Regenerate configuration C

When regenerating the configuration, the configuration data is read out of the PBX. The database is reset and the configuration data is restored to the PBX once more. This eliminates internal errors the configuration data may contain. The configuration of the PBX remains unchangend.



Further help under Backup

Remote Backup

Address of the backup server

Address of the server on which created backups are stored.

- Note: If available for the PBX, the Auerswald backup server is displayed here. If you want to use the Auerswald backup server, please contact your dealer or system partner.
- Further help under Backup

Password

Automatically generated, freely changeable password. Used to store a backup in a protected manner.

- Note: To change the current password, click on ⊚ Edit passwords.
 After you have entered your PIN or password, click on ⊚ Hide passwords.
- Caution: The backup can only be restored with knowledge of the associated password. If the password has been changed or deleted since the backup was created, the corresponding password must be entered to restore it. For this purpose, store the password in a safe place outside the PBX.
- Further help under <u>Backup</u>

Create backup 🕹

If a backup is created, the current configuration data is saved in a configuration file on an external server. The configuration files are archived with information about the time and date they were created.

Further help under Backup

Interval for automatic backups

off

Backups are not created automatically.

[Period]

Automatic backups are created regularly after the set period of time.

- Note: The time at which automatic backups are created depends on when the function is switched on. A subsequent change of the time period does not change the set time.
- Further help under Backup

Backups | Restoring 1

Restores the configuration data of the selected backup. The PBX restarts.

- Important: Restoring an old configuration overwrites the current configuration data.
- Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).

Backups | Delete selected backup X

Deletes the selected backup from the server.

Further help under Backup

Network settings

The network settings will also be restored.

The network settings include the following data:

- IP configuration (DHCP client setting, IP address, Sub-network mask, Gateway)
- DNS configuration (First DNS server, Second DNS server)
- HTTP proxy configuration (HTTP proxy settings)
- · Port of the web server
- Further help under Backup

Restoration points

Create Restoration Point , ↓,

If a restoration point for the configuration data is created, the current configuration data is saved in a configuration file in the PBX. The configuration files are archived with information about the time and date they were created.

Further help under <u>Backup</u>

Restoration point 1

Restores the configuration data of the selected restoration point. The PBX restarts.

- Important: Restoring an old configuration overwrites the current configuration data.
- Note: A configuration file that was created with an older firmware version, may possibly not contain all of the configuration data (for example, data for new functions that were added by a firmware update).
- Further help under <u>Backup</u>

Network settings

The network settings will also be restored.

The network settings include the following data:

 IP configuration (DHCP client setting, IP address, Sub-network mask, Gateway)

Restoration points

- DNS configuration (First DNS server, Second DNS server)
- HTTP proxy configuration (HTTP proxy settings)
- · Port of the web server
- Further help under Backup

Service data

Exporting service data 🗘

The PBX puts out service data that can be used for error analysis.

- Further help under <u>Backup</u>
- Further help under Service data

FTP directory

The service data is imported together with the FTP directory.

- **Note:** If the FTP directory is too full, exporting the service data may fail.
- Further help under Backup
- Further help under Service data

Update and Restart

Store firmware in the PBX
Firmware update via update server
Reset of the PBX (Restart)

Store firmware in the PBX

Manual Firmware Update

Starts the installation of a firmware file (*.rom) saved on a data storage device (PC). The new firmware is enabled after a restart. The execution of the necessary restart can immediately be enabled here or started separately at a later point of time.

Further help under Firmware Update

Firmware update via update server

Address of the update server

In the default settings: update.auerswald.de

Further help under <u>Firmware Update</u>

Online configuration

The PBX searches for available firmware updates on the update server. The availability is displayed when opening this page.

Further help under Firmware Update

Instant firmware update 🖓

Starts installation of a new firmware version from the update server.

Further help under Firmware Update

Firmware update, regular

Checks for available updates and their installation is exceeded according to the set interval and period of time.

None

No regular update takes place.

Further help under <u>Firmware Update</u>

Update Status

Displays information about the running and completed updates.

Firmware update via update server

Further help under Firmware Update

Reset of the PBX (Restart)

Restoring firmware (reboot) **

Restores the previous firmware (e.g. issues after an update).

- Caution: The PBX reboots automatically after you restore the previously used firmware. This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!
- Further help under <u>Firmware Update</u>

Reset of the PBX (Restart) C

Initiates a restart of the PBX after the set time.

- Caution: This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!
- Further help under Restart

Power down of the PBX

Initiates powering down of the PBX.

- ▲ Caution: This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!
- Further help under <u>Powering Down and Switching Off</u>

Message Configuration

Transfer of system e-mail Logs to be sent as e-mails Events for system logs Status Messages and Notes

Transfer of system e-mail

Outgoing mail server

Outgoing mail server that will be used to send system e-mails.

Default

System e-mails are sent by the default outgoing mail server.

- Further help under <u>E-mail</u>
- Further help under **System messages**

Send e-mail in HTML format

The system e-mails will be sent in HTML format.

- Note: When switched off, system e-mails are sent in text format.
- Further help under <u>E-mail</u>
- Further help under <u>System messages</u>

Language file

Language file whose text modules are to be used for system e-mails.

- Further help under E-mail
- Further help under <u>System messages</u>

URL format for hyperlinks

Format for the link in system e-mails to the system message overview.

with IPv4 address

Link starts with https: and the PBX's IPv4 address.

with IPv6 address

Link starts with https: and the PBX's IPv6 address (if present).

Custom

A URL prefix defined by the user is set.

- Further help under <u>E-mail</u>
- Further help under **System messages**

URL prefix

Network-specific part of the URL of user-defined URL format which comes before the device-specific part of the URL. This enables you to call hyperlinks from a telephone that is not on the PBX's internal network.

Example: Enter https://www.auerswald.de for this link https://www.auerswald.de/statics/html/index.htm#page_systemlogfiles.html

- Further help under <u>E-mail</u>
- Further help under <u>System messages</u>

Logo

Logo used on system e-mails.

- Further help under <u>E-mail</u>
- Further help under **System messages**

Message recipients

E-mail address to which system e-mails are sent. Up to 10 e-mail addresses can be entered here.

- Further help under E-mail
- Further help under System messages

Test

Send a test e-mail to the specified message recipients.

- Further help under E-mail
- Further help under **System messages**

Logs to be sent as e-mails

System Information

System information is sent as e-mails (except for system information about the sending of system e-mails).

- Note: When switched off, system information is only logged.
- Further help under <u>System messages</u>

Simple malfunction

Simple malfunctions are sent as e-mail.

- Note: When switched off, simple malfunctions are only logged.
- Further help under <u>System messages</u>

Serious malfunction

Serious malfunctions are sent as e-mails.

Further help under <u>System messages</u>

Security warning

Security warnings are sent as e-mails.

Further help under System messages

Waiting time until e-mail is sent

Specifies how long (in minutes) the PBX waits after the system event has occurred until it sends an e-mail. System events that occur during the waiting time are also included in this e-mail.



Further help under **System messages**

Events for system logs

Temperature monitoring | CPU alarm temperature

A serious malfunction is logged when the CPU temperature rises above this threshold.

Further help under **System messages**

Temperature monitoring | Power supply alarm temperature

A serious malfunction is logged when the power supply temperature rises above this threshold.

Further help under <u>System messages</u>

Relay monitoring

A system information log is created when a relay is switched on or off.

Further help under <u>System messages</u>

Status Messages and Notes

Reset hidden pop-up elements for | Execute

Warnings and notes that are hidden but still valid are reset for the selected user and shown as pop-up elements.



Further help under **System messages**

System messages

System messages

Each event is logged and displayed with detailled information in English.

- Note: When the maximum number of entries (max. 2.000 entries per category) is exceeded, old entries in the same category are automatically deleted.
- Further help under System messages

Selected filter

Filters the list of system messages depending on the category.

Further help under **System messages**

Monitoring and Overviews

External call channel assignment Dial plan

External call channel assignment

External call channel assignment

Display of the status as **free**, **in use** or**internal**. The display contains 20 dynamic VoIP channels for internal or external use.

- **Note:** When opening the page, it first shows the status which is automatically updated every 30 seconds.
- Further help under VolP

Dial plan

Filter

unfiltered

- with name: display of the freely selectable name in addition to the telephone number.
- without name: for a more compact overview only the telephone number is displayed.

General

- all utilized: telephone numbers assigned with a certain phone number type.
- all available: available telephone numbers.

according to the phone number type

The numbers of the selected phone number type are displayed.

Further help under <u>Internal numbers</u>

Configuring an entry

When clicking on a used telephone number (blue coloured), the configuration of the respective phone number type. Use

Back to get back to the dial plan from the configuration page.

Short Reference Phone

Information about the section
Using Different Telephone Types
Telephoning
Functions listed alphabetically

Information about the section

This section includes an alphabetical list of functions and settings of the PBX you can configure using an internal or, in some cases, an external phone.

The headlines explain which device (e.g. internal or external phone) may be used for the named character string.

Is the function also available to external devices, a remote switching number configured in the PBX has to be dialled first.

To simplify frequently used digit sequences on internal telephones, shortcode numbers can be used.



Further help under Short-code number

Using Different Telephone Types

Depending on the used phone type, there are differences in the operation and programming of the PBX. The following phones can be operated at the PBX:

Analogue Telephones (PD and DTMF)

If you are already in a call state (internal or external), you have to press the **Flash key (signal key, R key)** first on a DTMF phone before dialling a digit, e.g. for a call transfer.

Due to the difficulties concerning the transferring with PD phones, this function is not available for this kind of phones. If your telephone offers both dialling modes, you should therefore set it to DTMF.

Analogue T-Net Telephones

On some analogue DTMF telephones, specific T-Net functions can be performed in the analogue T-Net via the menu or via specific function keys. If you are operating such a phone as an internal phone, you can also use these comfortable operating modes for some functions of the PBX. Please note that the T-Net functions of the phone are limited because the PBX doesn't support some functions. If you want to use the T-Net functions of your phone, read the manual for the phone. When entering telephone numbers, take care to enter the leading exchange line access number. Instead of an "announcement of the central office", you will hear the confirmation tone for successful programming.

Standard VolP Telephones

On some standard VoIP phones, functions such as **Call waiting**, **Conference call**, **Query call** and **Splitting a call** are not carried out via

the PBX, but rather on the phone itself. Therefore, the instructions for operating these functions may deviate from the descriptions in this guide.

Normally, pressing the pound key # on a standard VoIP telephone initiates a dialling procedure. Since this key is required for numerous functions, as well as for all settings (programming), using the pound key for these actions is not possible on most standard VoIP phones (for more information on this, read the manual for the phone).

Telephoning

Incoming Calls
Call-Waiting Calls
Dialled calls
Unsuccessful outgoing calls
Query Calls
Transferring calls
Conference call

Incoming Calls

Accepting an incoming call

Pick up receiver.

Pick-up call (pick-up)

Description of operation under <u>Pick-up</u>

Performing call pick-up

Description of operation under <u>Pick-up</u>

Accepting door call and opening door

Description of operation under <u>Door functions</u>

Call-Waiting Calls

Rejecting a call-waiting call

Description of operation under <u>Call waiting</u>

Accepting a call-waiting call

Description of operation under <u>Call waiting</u>

Disconnecting the Current Call and Accepting the Call Waiting Call

Description of operation under <u>Call waiting</u>

Dialled calls

Initiating an Internal Call

Description of operation under <u>Dialled calls</u>

Initiating an External Call

Description of operation under <u>Dialled calls</u>

Initiating an External Call with Calling Line Identity Restriction

Description of operation under <u>Dialled calls</u>

Force Public Exchange Call for Automatic Exchange Line Request

Description of operation under <u>Force Public Exchange Call for</u>
Automatic Exchange Line Request

Initiating Targeted VolP Access

Description of operation under <u>Targeted VoIP access point</u>

Initiating door call and opening door

Description of operation under <u>Door functions</u>

Initiating loudspeaker audio out

Description of operation under <u>Loudspeaker Audio Out</u>

Unsuccessful outgoing calls

Initiating recall on busy/no reply

- Note: The PBX does not support the recall function with analogue telephones.
- To do this with a VoIP phone, please refer to the instructions for the telephone.

Initiating a priority call when Do-not-Disturb is enabled

10 seconds.

Query Calls

Initiating second call

Description of operation under Query and Transfer

Ending a query call

Description of operation under <u>Query and Transfer</u>

Splitting a call

Description of operation under <u>Query and Transfer</u>

Ending one of the two calls

Description of operation under <u>Query and Transfer</u>

Connecting the two call partners to each other

Description of operation under <u>Query and Transfer</u>

Initiating a query call to third communication partner

Description of operation under Query and 3rd Call

Transferring calls

Transfer the call with an announcement

Description of operation under <u>Query and Transfer</u>

Transferring without Announcement

Description of operation under <u>Query and Transfer</u>

Calling a third communication partner and transferring to an active communication partner

Description of operation under <u>Query and Transfer</u>

Putting the Call into the Internal Parking Zone (Call Parking)

Description of operation under <u>Call on hold</u>

Picking up the Call from the Internal Parking Zone (Call Parking)

Description of operation under <u>Call on hold</u>

Conference call



Further help under Conference rooms

Functions listed alphabetically

Announcement before answering

Automatic reception

Blocklist (incoming)

Call Data (Single Call Record)

Call Forwarding for external numbers

Call Forwarding for groups

Call Forwarding for users

Call on hold

Call Through

Call waiting

Conference rooms

Configuration switchover

Date and Time

Dialled calls

Do-not-Disturb

Door functions

Emergency Call Announcement

Firmware Update

Flash time

FMC

Follow-me

Force Public Exchange Call for Automatic Exchange Line Request

Groups

Identification (IP Address, Subnet Mask, Gateway, DHCP Client)

InterCom

LAN TAPI

Loudspeaker Audio Out

Multi-path call forwarding

Functions listed alphabetically

Music on Hold and Announcement
Pick-up
PIN
Query and 3rd Call
Query and Transfer
Relays
Roaming User
Targeted VoIP access point
Voice mailbox
Allowlist (incoming)

Announcement before answering

Recording an Announcement (on an internal phone)

8 > PIN > * 56 > Announcement > * > Action > * > Action >

PIN Admin PIN

Announcement 01 ... 10 (no. 1-10 of the announcement)

Action Wait for trilling tone after 2nd *.

Start recording with 3rd *.

Stop recording with #.

Afterwards the recording is played back for checking.

Further help under Announcement before answering

Deleting the Announcement (on an internal phone)

8 > PIN > * 55 > Announcement >

PIN Admin PIN

Announcement 01 ... 10 (no. 1-10 of the announcement)

Further help under Announcement before answering

Listening to the Announcement (on an internal phone)

8 > PIN > * 57 > Announcement >

PIN Admin PIN

Announcement 01 ... 10 (no. 1-10 of the announcement)

Further help under Announcement before answering

Automatic reception

Recording an Announcement (on an internal phone)

8 > PIN > * 564 > Announcement > * > Action > * > Action >

PIN Admin PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Action Wait for trilling tone after 2nd *.

Start recording with 3rd *.

Stop recording with #.

Afterwards the recording is played back for checking.

Further help under <u>Automatic reception</u>

Deleting the Announcement (on an internal phone)

8 > PIN > * 554 > Announcement >

PIN Admin PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Further help under <u>Automatic reception</u>

Listening to the Announcement (on an internal phone)

8 > PIN > * 574 > Announcement >

Automatic reception

PIN	Admin PIN
Announcement	1 9, 0 (no. 1-10 of the announcement)

Further help under <u>Automatic reception</u>

Blocklist (incoming)

Enabling blocklist (incoming) for users (on an internal phone)



PIN User PIN

Further help under Block- and allowlist

Disabling blocklist (incoming) for users (on an internal phone)

8 > PIN > * 23 0

PIN User PIN

Further help under Block- and allowlist

Call Data (Single Call Record)

Deleting Own Call Data Records (Works Council Function - on an internal phone)



8 > PIN > * 51

PIN

User PIN

Additional information Authorisation required

Further help under Call data

Call Forwarding for external numbers

Configuring unconditional call forwarding (on an internal phone)

96 > Own number > # * 21 * > Destination number >

Own number	Own phone number (number without exchange line access number), if call forwarding is to be configured for a specific number.
	The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line
Additional information	Authorisation required
	Call forwarding for external numbers can be configured for the current configuration.

Further help under Call forwarding

Enabling unconditional call forwarding without re-entering the number (on an internal phone)



96 > Own number > # * 21

Own number	Own phone number (number without exchange line access number), if call forwarding is to be configured for a specific number.
	The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required Call forwarding for external numbers can be configured for the current configuration.

Further help under Call forwarding

Disabling unconditional call forwarding (on an internal phone)

	## 96 > Own number > ## 21 #
--	------------------------------

Own number	Own phone number (number without exchange line access number), if call forwarding is to be configured for a specific number.
	The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Additional information	Authorisation required

Further help under Call forwarding

Configuring call forwarding on busy (on an internal phone)

96 > Own number > # * 67 * > Destination number >

Own number Own phone number (number without exchange

line access number), if call forwarding is to be

configured for a specific number.

The number can be omitted if call forwarding is to be configured for the number currently sent by

the phone.

Destination number External phone to be called

> External number (for exchange line request type Internal phone: with preceding exchange line

access number)

Additional information Authorisation required

Further help under <u>Call forwarding</u>

Enabling call forwarding on busy without re-entering the number (on an internal phone)



96 > Own number > # * 67 # >

Own number Own phone number (number without exchange

line access number), if call forwarding is to be

configured for a specific number.

The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.

Additional information Authorisation required

Further help under <u>Call forwarding</u>

Disabling call forwarding on busy (on an internal phone)

96 > Own number > ## 67

line access number), if call forwarding is to be

configured for a specific number.

The number can be omitted if call forwarding is to be configured for the number currently sent by

the phone.

Additional information Authorisation required

Further help under <u>Call forwarding</u>

Configuring call forwarding on no reply (on an internal phone)

96 > Own number > # * 61 * > Destination number >

line access number), if call forwarding is to be

configured for a specific number.

	The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation required

Further help under <u>Call forwarding</u>

Enabling call forwarding on no reply without re-entering the number (on an internal phone)

96 > Own number > # * 61

Own number

Own phone number (number without exchange line access number), if call forwarding is to be configured for a specific number.

The number can be omitted if call forwarding is to be configured for the number currently sent by the phone.

Additional information Authorisation required

Further help under <u>Call forwarding</u>

Disabling call forwarding on no reply (on an internal phone)

96 > Own number > ## 61

Own number Own phone number (number without exchange

line access number), if call forwarding is to be

configured for a specific number.

The number can be omitted if call forwarding is to be configured for the number currently sent by

the phone.

Additional information Authorisation required

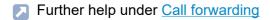
Further help under Call forwarding

Call Forwarding for groups

Configuring unconditional call forwarding (on an internal phone)

97 > Group number > # * 21 * > Destination number >

Group telephone number	Internal group number The number can be omitted if the phone is logged in to the group as outgoing.
Destination number	Internal number of the group or user or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group required



Enabling unconditional call forwarding without re-entering the number (on an internal phone)



97 > Group number > # * 21

Group telephone number

Internal group number

Call Forwarding for groups

	The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Further help under <u>Call forwarding</u>

Disabling unconditional call forwarding (on an internal phone)

II ## 97 > Group number > ## 21 #
--

Group telephone	Internal group number
number	The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Further help under <u>Call forwarding</u>

number

Configuring call forwarding on busy (on an internal phone)

## 97 > Group nun	nber > # * 67 * > Destination number > #
Group telephone	Internal group number

The number can be omitted if the phone is logged in to the group as outgoing.

	or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group required

Further help under Call forwarding

Enabling call forwarding on busy without re-entering the number (on an internal phone)

97 > Group number > # * 67

Group telephone number	Internal group number
	The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Further help under <u>Call forwarding</u>

Disabling call forwarding on busy (on an internal phone)

97 > Group number > ## 67

Group telephone number	Internal group number
	The number can be omitted if the phone is logged in to the group as outgoing.
Additional information	Authorisation and membership of the group required

Further help under Call forwarding

Configuring call forwarding on no reply (on an internal phone)

97 > Group number > # * 61 * > Destination number >

Group telephone number	Internal group number The number can be omitted if the phone is logged in to the group as outgoing.
Destination number	Internal number of the group or user or External number (for exchange line request type Internal phone: with preceding exchange line access number)
Additional information	Authorisation for external destination required Authorisation and membership of the group required

Further help under Call forwarding

Enabling call forwarding on no reply without re-entering the number (on an internal phone)

97 > Group number > # * 61

Group telephone Internal group number

number The number can be omitted if the phone is

logged in to the group as outgoing.

Additional information Authorisation and membership of the group

required

Further help under <u>Call forwarding</u>

Disabling call forwarding on no reply (on an internal phone)

97 > Group number > ## 61

Group telephone Internal group number

number The number can be omitted if the phone is

logged in to the group as outgoing.

Authorisation and membership of the group Additional information

required

Further help under Call forwarding

Call Forwarding for users

Configuring unconditional call forwarding (on an internal phone)

* 21 * > Destination number > #

Internal number of the group or user

or

External number (for exchange line request type Internal phone: with preceding exchange line

access number)

Additional information Authorisation for external destination required

Further help under Call forwarding

Enabling unconditional call forwarding without re-entering the number (on an internal phone)

- * 21 #
- Further help under <u>Call forwarding</u>

Disabling unconditional call forwarding (on an internal phone)

- # 21 #
- Further help under <u>Call forwarding</u>

Configuring call forwarding on busy (on an internal phone)

* 67 * > Destination number > #

Internal number of the group or user

or

External number (for exchange line request type Internal phone: with preceding exchange line access number)

Additional information Authorisation for external destination required

Further help under <u>Call forwarding</u>

Enabling call forwarding on busy without re-entering the number (on an internal phone)

- * 67 #
- Further help under Call forwarding

Disabling call forwarding on busy (on an internal phone)

- # 67 #
- Further help under Call forwarding

Configuring call forwarding on no reply (on an internal phone)

* 61 * > Destination number > #

Internal number of the group or user

or

External number (for exchange line request type Internal phone: with preceding exchange line access number)

Additional information Authorisation for external destination required

Further help under Call forwarding

Enabling call forwarding on no reply without re-entering the number (on an internal phone)

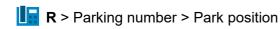
- * 61 #
- Further help under Call forwarding

Disabling call forwarding on no reply (on an internal phone)

- # 61 #
- Further help under Call forwarding

Call on hold

Putting a Call into the Parking Zone (on an internal phone)



R	Flash key or Query key
Parking number	Internal number of the parking zone
park position	0 9 (no. 0-9 of the park position)
Additional information	Authorisation required

Further help under Call on hold

Taking over a Call/Conversation (on an internal phone)

Parking number > Park position

Parking number	Internal number of the parking zone
park position	0 9 (no. 0-9 of the park position)
Additional information	Authorisation required

Further help under Call on hold

Taking over a Call/Conversation in an Ongoing Call (on an internal phone)



R > ## 06 > Parking number > Park position

R	Flash key or Query key
Parking number	Internal number of the parking zone
park position	0 9 (no. 0-9 of the park position)
Additional information	Authorisation required



Further help under Call on hold

Call Through

Initiating a Call (on an external phone)

Call Through number > PIN > * > Destination number

Call Through number	External number defined for Call Through on the PBX
PIN	Call Through PIN configured in the PBX
Destination number	Internal or external phone to be called Internal number of the group or user External number
Additional information	# after the destination number accelerates the dialling

Further help under Call Through

Call waiting

Rejecting a Knocking Caller (on an Analogue internal phone)



R Flash key or Query key

Further help under Call waiting

Accepting a Knocking Caller and Disconnecting the Active Call (on an Analogue internal phone)

R > 1

R Flash key or Query key

Further help under <u>Call waiting</u>

Accepting a Knocking Caller and Putting the Active Call on Hold (on an Analogue internal phone)

I R > 2

R Flash key or Query key

Further help under <u>Call waiting</u>

Conference rooms

Recording an Announcement (on an internal phone)

8 > PIN > * 562 > Announcement > * > Action > * > Action >

PIN Admin PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Action Wait for trilling tone after 2nd *.

Start recording with 3rd *.

Stop recording with #.

Afterwards the recording is played back for checking.

Further help under Conference rooms

Deleting the Announcement (on an internal phone)

8 > PIN > * 552 > Announcement >

PIN Admin PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Further help under Conference rooms

Listening to the Announcement (on an internal phone)

8 > PIN > * 572 > Announcement >

Conference rooms

PIN Admin PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Further help under Conference rooms

Configuration switchover

Switching the Configuration (on an internal phone)

8 > * 81 > Configuration >

Configuration 2- to 4-digit ID no. of the configuration

Additional information Authorisation required

Further help under <u>Configuration switchover</u>

Switching the Configuration (on an external phone)

Remote switching number > ## 8 > PIN > * 81 > Configuration > #

Remote switching Remote switching number for remote programming

PIN User PIN

Configuration 2- to 4-digit ID no. of the configuration

Additional information Authorisation required

- Further help under External access
- Further help under Configuration switchover

Date and Time

Setting the Time (on an internal phone)



PIN	Admin PIN
hh	00 23 (hour)
mm	00 59 (minute)
SS	00 59 (second)

Further help under Date and Time

Setting the Date (on an internal phone)

8 > PIN > * 301 > dd > mm > yy >

PIN	Admin PIN
dd	01 31 (day)
mm	01 12 (month)
уу	00 99 (year)

Further help under Date and Time

Dialled calls

Initiating an Internal Call (on an internal phone)

Destination number

Destination Internal phone to be called

number Internal number of the group or user

Further help under <u>Dialled calls</u>

Initiating an External Call (on an internal phone)

Destination number

Destination External phone to be called

number External number (for exchange line request type

Internal phone: with preceding exchange line access

number)

Further help under <u>Dialled calls</u>

Initiating an External Call with Suppressed Number Presentation (on an internal phone)

📗 * **31 #** > Destination number

Destination External phone to be called

number

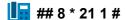
Dialled calls

External number (for exchange line request type Internal phone: with preceding exchange line access number)

Further help under Dialled calls

Do-not-Disturb

Enabling Do-not-Disturb (on an internal phone)



Further help under **Do-not-Disturb**

Disabling Do-not-Disturb (on an internal phone)



Further help under **Do-not-Disturb**

Door functions

Initiating a Door Call (on an internal phone)

Internal door number

Further help under **Door functions**

Opening a Door (on an internal phone)

I R > #

R Flash key or Query key

Additional Authorisation required information

Further help under **Door functions**

Initiating a Query Call from a Door Call (on an Analogue internal phone)

R > Internal door number

R Flash key or Query key

Phone number Internal number of the group or user or

Door functions

External number (for exchange line request type Internal phone: with preceding exchange line access number)

Further help under **Door functions**

Emergency Call Announcement

Recording an Announcement (on an internal phone)

8 > PIN > * 5650 > * > Action > * > Action >

PIN Admin PIN

Action Wait for trilling tone after 2nd *.

Start recording with 3rd *.

Stop recording with #.

Afterwards the recording is played back for checking.

Further help under **Emergency calls**

Listening to the Announcement (on an internal phone)



PIN Admin PIN

Further help under **Emergency calls**

Deleting the Announcement (on an internal phone)

8 > PIN > * 5550

PIN Admin PIN

Further help under Emergency calls

Firmware Update

Initiating an automatic PBX update (on an internal phone)



8 > PIN > * 92 0

PIN Admin PIN Additional information First end all calls, then initiate update Authorisation required

Further help under Firmware Update

Flash time

Learning the Flash Timing (on an Analogue internal phone)



R Flash key or Query key

Further help under Query and Transfer

FMC

Switching on the calling of the user's mobile telephones (on an internal phone)



PIN User PIN

Additional information The user must be configured as the mobile telephones' owner.

Further help under FMC

Switching off the calling of the user's mobile telephones (on an internal phone)



PIN User PIN

Additional information The user must be configured as the mobile telephones' owner.

Further help under FMC

Switching on the calling of the user's mobile telephones (on an external phone)

Remote switching number > ## 8 > PIN > * 18 1 #

PIN	User PIN
Remote switching number	Remote switching number for remote programming
Additional information	After entering the remote switching number, wait for external confirmation signal (1 second). Continue with DTMF entry.
	The user must be configured as the mobile telephones' owner.

- Further help under <u>External access</u>
- Further help under <u>FMC</u>

Switching off the calling of the user's mobile telephones (on an external phone)

Remote switching number > ## 8 > PIN > * 18 0 #

PIN	User PIN
Remote switching number	Remote switching number for remote programming
Additional information	After entering the remote switching number, wait for external confirmation signal (1 second). Continue with DTMF entry.
	The user must be configured as the mobile telephones' owner.

- Further help under External access
- Further help under <u>FMC</u>

Initiating a call (on the user's mobile telephone)

Call Through number > PIN > * > Destination number

Call Through number	External number defined for Call Through on the PBX
PIN	User PIN
Destination number	Internal or external phone to be called Internal number of the group or user External number
Additional information	# after the destination number accelerates the dialling The user must be assigned to the mobile telephone or configured as its owner.

Further help under FMC

Follow-me

Enabling Follow-me (on the Internal Destination Phone)

8 * 19 1 > User/group number >

User/group number Phone/group to be redirected

Internal number of the group or user

Additional information Authorisation required

Further help under Follow-me

Disabling Follow-me (on the Internal Destination Phone)

Deletes all redirections via Follow-me with this phone as destination.



8 * 19 0

Additional information Authorisation required

Further help under Follow-me

Enabling Follow-me (on any internal phone)



8 * 19 1 > User/group number > * > Destination number >

User/group number Phone/group to be redirected

Internal number of the group or user

Destination number Internal or external phone to be called Internal number of the group or user
or
External number (for exchange line request type
Internal phone: with preceding exchange line
access number)

Additional information Authorisation required

For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply for the user are proved before a Follow-me is executed.

Further help under Follow-me

Disabling Follow-me (on any internal phone)

III ## 8 * 19 0 > User/group number > #

User/group number Redirected phone/group

Internal number of the group or user

Additional information Authorisation required

Further help under Follow-me

Enabling Follow-me (on an external phone)

Remote switching number > ## 8 > PIN > * 19 1 > User/group number * > Destination number > #

PIN	User PIN of the forwarding user
Remote switching number	Remote switching number for remote programming
User/group number	Phone/group to be redirected Internal number of the group or user
Destination number	Internal or external phone to be called Internal number of the group or user or External number
Additional information	After entering the remote switching number, wait for external confirmation signal (1 second). Continue with DTMF entry. Authorisation required For call forwarding to an external number: The external destination must be reachable for the user. For this, the exchange line routing settings (with block- and allowlist) that apply for the user are proved before a Follow-me is executed.

- Further help under External access
- Further help under Follow-me

Disabling Follow-me (on an external phone)

Remote switching number > ## 8 > PIN > * 19 0 > User/group number > #

PIN	User PIN of the forwarding user
Remote switching number	Remote switching number for remote programming
User/group number	Phone/group to be redirected Internal number of the group or user
Additional information	After entering the remote switching number, wait for external confirmation signal (1 second). Continue with DTMF entry. Authorisation required

- Further help under <u>External access</u>
- Further help under Follow-me

Force Public Exchange Call for Automatic Exchange **Line Request**

Note: This function is necessary only in certain countries (such as France, Belgium), if certain special numbers are dialled.

Force Public Exchange Call for Automatic Exchange Line Request



94 > Destination number

Destination External phone to be called

number External number (without preceding exchange line

access number)

Further help under Exchange line request

Groups

Logging in to a Group (Incoming and Outgoing - on an internal phone)

8 * 40 1 > Group number >

Group telephone Internal group number number

Further help under Groups

Logging out of a Group (incoming and outgoing - on an internal phone)

8 * 40 0 > Group number >

Group telephone Internal group number number

Further help under Groups

Logging out of All Groups (incoming and outgoing - on an internal phone)

- ## 8 * 40 0 #
- Further help under Groups

Logging in to a Group (only incoming - on an internal phone)

8 * 41 1 > Group number >

Group telephone number

Internal group number

Further help under Groups

Logging in to All Groups (only incoming - on an internal phone)

- ## 8 * 41 1 #
- Further help under Groups

Logging out of a Group (only incoming - on an internal phone)

8 * 41 0 > Group number >

Group telephone number

Internal group number

Further help under Groups

Logging out of All Groups (only incoming - on an internal phone)

- ## 8 * 41 0 #
- Further help under Groups

Logging in to a Group (only outgoing - on an internal phone)

8 * 42 1 > Group number >

Group telephone Internal group number number

Further help under Groups

Logging out of the Group (only outgoing - on an internal phone)

- ## 8 * 42 0 #
- Further help under Groups

Logging all users out of a group (incoming and outgoing - on an internal phone)

8 * 48 0 > Group number >

Group telephone Internal group number number

Additional information Membership of the group required

Further help under Groups

Identification (IP Address, Subnet Mask, Gateway, **DHCP Client)**

- Note: Only IPv4 addresses can be entered using a character string on the phone. IPv6 addresses must be entered using the web interface of the PBX.
- Note: These settings may not be applicable to a configured second network interface.

Configuring the IPv4 Address of the PBX (on an internal phone)



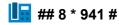
8 > PIN > * 931 * > IP address >

PIN Admin PIN

IPv4 address of the PBX (12 digits) IP address

Further help under Network

Querying the IPv4 Address of the PBX (on an internal phone)



Additional information Display of the IPv4 address of the PBX on the phone display after callback by the PBX

Further help under <u>Network</u>

Configuring the Subnet Mask of the PBX (on an internal phone)

8 > PIN > * 932 * > Subnet mask >

PIN Admin PIN

Subnet mask Subnet mask (12 digits)

Further help under Network

Querying the Subnet Mask of the PBX (on an internal phone)

8 * 942

Additional information Display of the subnet mask on the phone display after callback by the PBX

Further help under Network

Configuring the Gateway of the PBX (on an internal phone)

8 > PIN > * 933 * > IP address >

PIN Admin PIN

IP address IPv4 address of the gateway (12 digits)

Further help under Network

Querying the Gateway of the PBX (on an internal phone)

8 * 943

Additional information Display of the IPv4 address of the gateway on the phone display after callback by the PBX

Further help under Network

Switching the DHCP Client of the PBX on (on an internal phone)

8 > PIN > * 930 * 1

PIN Admin PIN

Further help under Network

Switching the DHCP Client of the PBX off (on an internal phone)

8 > PIN > * 930 * 0

PIN Admin PIN

Further help under Network

Querying the DHCP Client Status of the PBX (on an internal phone)

8 * 940

Additional information Display of the DHCP client status of the PBX on the phone display after callback by the PBX

0 = off

1 = on



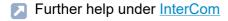
InterCom

Running InterCom on a phone number (on an internal telephone)



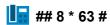
011 > Destination number

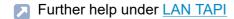
Destination number	Internal number of the group or user
Additional information	Authorisation required on the destination telephone



LAN TAPI

Set telephone as TAPI telephone (on the internal telephone concerned)





Loudspeaker Audio Out

Announcement to Announcement Output (on an internal phone)

Number of the announcement output

Number of the Internal number of the announcement output announcement output

Further help under <u>Loudspeaker Audio Out</u>

Multi-path call forwarding

Configuring multi-path call forwarding for users (on an internal phone)

* 481 * > Destination number > #

Destination number

Internal or external phone to be called

Internal number of the group or user

or

External number (for exchange line request type

Internal phone: with preceding exchange line access

number)

Further help under Multi-path call forwarding

Enabling Multi-Path call forwarding for users (on an internal phone)

* 481 #

Further help under Multi-path call forwarding

Disabling Multi-Path call forwarding for users (on an internal phone)

481

Further help under Multi-path call forwarding

Check activation (user - on an internal phone)

* # 481 #

Additional information Busy signal = multi-path call forwarding is disabled

Further help under Multi-path call forwarding

Configuring Multi-Path call forwarding for Groups (on an internal group phone)

97 > Group number > # * 481 * > Destination number >

Group telephone number

Destination number

Internal or external phone to be called Internal number of the group or user or

External number (for exchange line request type Internal phone: with preceding exchange line access number)

Additional information

Membership of the group required

Further help under Multi-path call forwarding

Enabling multi-path call forwarding for Groups (on an internal group phone)

97 > Group number > # * 481 *

Group telephone

Internal group number

number

Additional information Membership of the group required

Further help under Multi-path call forwarding

Disabling Multi-Path call forwarding for Groups (on an internal group phone)



97 > Group number > ## 481

Group telephone

Internal group number

number

Additional information Membership of the group required

Further help under Multi-path call forwarding

Check activation (group - on an internal phone)



97 > Group number > # * # 481

Group telephone

Internal group number

number

Additional information Busy signal = multi-path call forwarding is disabled

Further help under <u>Multi-path call forwarding</u>

Music on Hold and Announcement

Recording Music on Hold at Music Input (on an internal phone)

8 > PIN > * 5632 > * > Action > * > Action >

PIN Admin PIN

Action Wait for trilling tone after 2nd *.

Start recording with 3rd * and simultaneously switch on music equipment.

Stop recording with #.

Further help under Music on Hold and Announcement

Deleting Music on Hold (on an internal phone)

8 > PIN > * 5532

PIN Admin PIN

Further help under Music on Hold and Announcement

Recording an Announcement (on an internal phone)

8 > PIN > * 5631 > * > Action > * > Action >

PIN Admin PIN

Action Wait for trilling tone after 2nd *.

Start recording with 3rd *.

Stop recording with #.

Further help under Music on Hold and Announcement

Deleting the Announcement (on an internal phone)



PIN Admin PIN

Further help under Music on Hold and Announcement

Switching on/off music on hold and announcement and listening to it (on an internal phone)

8 > PIN > * 573 > Option >

PIN	Admin PIN
option	0 (no music on hold/announcement)
	1 (announcement only)
	2 (Music on Hold only)
	3 (Music on hold + announcement)

Further help under Music on Hold and Announcement

Pick-up

Taking over a Call/Conversation (on an internal phone)



06 > User/group number

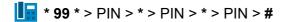
User/group number Internal number of the group or user

Additional information Authorisation required

Further help under Pick-Up and Call Pick-Up

PIN

Changing the PIN (on an internal phone)



1st PIN	Old PIN to be changed = Admin or user PIN
2nd PIN	New PIN = Admin or user PIN
3rd PIN	New PIN = Admin or user PIN

- Note: If, at the end, you hear a busy signal instead of a confirmation tone, the new PIN may already be in use in the PBX and no change takes place. Re-attempt with another digit sequence.
- Further help under <u>Access authorisations</u>

Query and 3rd Call

Initiating a 3rd Call (on an Analogue internal phone)



R > **5** > Destination number

R	Flash key or Query key
Destination number	Internal or external phone to be called Internal number of the group or user
	or External number (for exchange line request type Internal phone: with preceding exchange line access number)

Further help under **Query and Transfer**

Disconnecting the 3rd Call and Returning to the Last Active Call (on an Analogue internal phone)



R Flash key or Query key

Further help under Query and Transfer

Disconnecting the 3rd Call and Returning to the Last Call on Hold (on an Analogue internal phone)



R Flash key or Query key

Further help under **Query and Transfer**

Connecting 2nd and 3rd Call Partner and Continuing to Talk to the 1st Call Partner (on an Analogue internal phone)

R > 4

R Flash key or Query key

Further help under Query and Transfer

Continuing the 3rd Call and Disconnecting the Other Calls (on an Analogue internal phone)

R > 0

R Flash key or Query key

Further help under Query and Transfer

Query and Transfer

Initiating a Query (on an Analogue internal phone)



R	Flash key or Query key
Destination number	Internal or external phone to be called Internal number of the group or user
	or External number (for exchange line request type Internal phone: with preceding exchange line access number)

Further help under <u>Query and Transfer</u>

Cancelling a Ringing Call (on an Analogue internal phone)



R	Flash key or Query key	
---	------------------------	--

Further help under **Query and Transfer**

Splitting (on an Analogue internal phone)



R Flash key or Query key

Further help under Query and Transfer

Disconnecting a Call on Hold in the Background (on an Analogue internal phone)

R > 0

R Flash key or Query key

Further help under Query and Transfer

Disconnecting the Active Call and Returning to the user on Hold (on an Analogue internal phone)

I R > 1

R Flash key or Query key

Further help under Query and Transfer

Connecting Call Partners (on an Analogue internal phone)

I R > 4

R Flash key or Query key

Further help under Query and Transfer

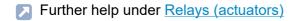
Relays

Switching a Relay on - Switches off Automatically after Configured Switching Duration (on an internal phone)

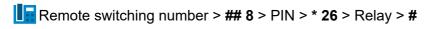


8 * 26 > Relay >

Relays	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes light, door opener and PC relay
	Authorisation required



Switching a Relay on - Switches off Automatically after Configured **Switching Duration (on an external phone)**



Remote switching number	Remote switching number for remote programming
PIN	User PIN
Relays	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes light, door opener and PC relay Authorisation required

Relays

- Further help under <u>External access</u>
- Further help under Relays (actuators)

Switching a Relay on (on an internal phone)

8 * 26 > Relay > * 1

Relays 2- to 4-digit ID no. of the relay

Additional information For relays with the operating modes manual

relay, configuration-dependent and automatic

configuration switchover

Authorisation required

Further help under Relays (actuators)

Switching a Relay on (on an external phone)

Remote switching number > ## 8 > PIN > * 26 > Relay > * 1 #

Remote switching number	Remote switching number for remote programming
PIN	User PIN
Relays	2- to 4-digit ID no. of the relay
Additional information	For relays with the operating modes manual relay, configuration-dependent and automatic configuration switchover

Authorisation required

- Further help under <u>External access</u>
- Further help under Relays (actuators)

Switching a Relay off (on an internal phone)

8 * 26 > Relay > * 0

Relays 2- to 4-digit ID no. of the relay

Additional information For relays with the operating modes manual

relay, configuration-dependent and automatic

configuration switchover

Authorisation required

Further help under Relays (actuators)

Switching a Relay off (on an external phone)

Remote switching number > ## 8 > PIN > * 26 > Relay > * 0 #

Remote switching number for remote

number programming

PIN User PIN

Relays 2- to 4-digit ID no. of the relay

Relays

Additional information For relays with the operating modes manual

relay, configuration-dependent and automatic

configuration switchover

Authorisation required

Further help under <u>External access</u>

Further help under Relays (actuators)

Roaming User

Logging on to a telephone (on the affected internal phone)



8 > PIN > * 62 1 > user phone number >

PIN	Admin or user PIN
User phone number	Internal user number
Additional information	Logging in automatically triggers logging out on the telephone that is currently being used.
	The user must either be configured as the telephone's user or have authorisation as a roaming user.

Further help under Roaming User

Logging out of a telephone (on the affected internal phone)



8 * 61 0

Additional information The user must either be configured as the telephone's user or have authorisation as a roaming user. When they log out, the local caller list on a COMfortel D-series telephone is deleted.

Further help under Roaming User

Logging on to an additional telephone (on the affected internal phone)



8 > PIN > * 61 1 > user phone number >

PIN	Admin or user PIN
User phone number	Internal user number
Additional information	The user must either be configured as the telephone's user or have authorisation as a roaming user.



Further help under Roaming User

Logging out of all telephones(on an affected internal phone)



8 * 62 0

Additional information

The user must either be configured as the telephone's user or have authorisation as a roaming user.

When they log out, the local caller list on a COMfortel D-series telephone is deleted.

Further help under Roaming User

Targeted VoIP access point

Initiating an External Call Using Targeted VolP Access (on an internal phone)



** > Account number > Destination number

Account number	Used VoIP account Exchange line access number (account number)
Destination number	External phone to be called External number (for exchange line request type Internal phone: with preceding exchange line
	access number)
Additional information	Only for exchange line request type Internal phone: ** is not necessary

- Further help under Dialled calls
- Further help under Exchange line request

Voice mailbox

Disabling the Readiness (on an internal phone If the internal phone Is the Owner of the Voice Mailbox)

- ## 8 * 28 * 0 #
- Further help under Voice mailbox

Enabling the Readiness (on an internal phone if the internal phone is the Owner of the Voice Mailbox)

- ## 8 * 28 * 1 #
- Further help under Voice mailbox

Enabling the Time Control for Readiness (on an internal phone If the internal phone Is the Owner of the Voice Mailbox)

- ## 8 * 28 * 9 #
- Further help under Voice mailbox

Recording an Announcement for a Voice Mailbox with a Phone (on an internal phone)

8 > PIN > * 566 > Announcement > Voice mailbox number > * > Action > * > Action >

PIN Admin or user PIN

Announcement 1 ... 9, 0 (no. 1-10 of the announcement)

Voice mailbox

Voice mailbox number	Internal number of the voice mailbox
Action	Wait for trilling tone after 2nd *. Start recording with 3rd *. Stop recording with #. Afterwards the recording is played back for checking.

Further help under Voice mailbox

Listening to the Announcement for a Voice Mailbox (on an internal phone)



PIN	Admin or user PIN
Announcement	1 9, 0 (no. 1-10 of the announcement)
Voice mailbox number	Internal number of the voice mailbox

Further help under Voice mailbox

Deleting the Announcement for a Voice Mailbox (on an internal phone)



PIN Admin or user PIN

Voice mailbox

Announcement	1 9, 0 (no. 1-10 of the announcement)
Voice mailbox number	Internal number of the voice mailbox

Further help under Voice mailbox

Querying a Voice Mailbox (on an internal phone If the internal phone Is the Owner of the Voice Mailbox)

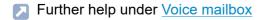


Voice mailbox number > Function

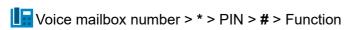
Voice mailbox number	Internal number of the voice mailbox
Function	0 (Opens the Help menu.)
	1 (Opens the main menu.)
	2 (Plays back new messages.)
	3 (Plays back all of the messages.)
	4 (Jumps back to the previous message.)
	5 (Repeats the message currently being played back.)
	6 (Jumps to the next message.)
	7 (Deletes the message currently being played back.)
	8 (Ends playback.)
	* 10 (Disables the voice mailbox.)
	* 11 (Enables the voice mailbox.)

	* 20 (Enables recording and changing announcements.)
Additional information for * 20	The default announcement and your announcements are played, if present. The following functions are available whilst your own announcements are being played back:
	# (Selects the currently heard announcement as active announcement.)
	4 (Jumps to the previous announcement.)
	5 (Repeats the announcement you have just heard.)
	6 (Jumps to the next announcement.)
	7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)
Function	* 30 (Disables message forwarding.)
	* 31 (Enables message forwarding.)
	* 70 (Deletes all messages and voice memos.)
	* 99 (Records a voice memo.)
	Hang-up (Ends remote access.)
	Additional function while a message is being replayed:
	9 (Announces the phone number of the caller – provided it has been transferred.)

	Further options after announcement of the phone number:
	# (Initiates a callback.) * (Back to the message playback.)
Additional information	After having dialled the internal number, wait until the announcement has ended



Querying a Voice Mailbox (on an internal phone If the internal phone Is not the Owner of the Voice Mailbox and Readiness of the Voice Mailbox Has Been Enabled)



Voice mailbox number	Internal number of the voice mailbox
PIN	User or remote access PIN
Function	0 (Opens the Help menu.)
	1 (Opens the main menu.)
	2 (Plays back new messages.)
	3 (Plays back all of the messages.)
	4 (Jumps back to the previous message.)
	5 (Repeats the message currently being played back.)
	6 (Jumps to the next message.)

	7 (Deletes the message currently being played back.)
	8 (Ends playback.)
	* 10 (Disables the voice mailbox.)
	* 11 (Enables the voice mailbox.)
	* 20 (Enables recording and changing announcements.)
Additional information for * 20	The default announcement and your announcements are played, if present. The following functions are available whilst your own announcements are being played back:
	# (Selects the currently heard announcement as active announcement.)
	4 (Jumps to the previous announcement.)
	5 (Repeats the announcement you have just heard.)
	6 (Jumps to the next announcement.)
	7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)
Function	* 30 (Disables message forwarding.)
	* 31 (Enables message forwarding.)
	* 70 (Deletes all messages and voice memos.)
	* 99 (Records a voice memo.)

Hang-up (Ends remote access.)

Additional function while a message is being replayed:

9 (Announces the phone number of the caller – provided it has been transferred.)

Further options after announcement of the phone number:

(Initiates a callback.)

* (Back to the message playback.)

Additional information After having dialled the internal number, press *

while the announcement is being played.

After having pressed #, wait until the

announcement has ended

Further help under Voice mailbox

Remote Access (on an External Phone If Readiness of the Voice Mailbox Is Enabled)

IVoice mailbox number > * > PIN > **#** > Function

Voice mailbox number External number of the voice mailbox

or

External number of the phone which is owner/

user of the voice mailbox

PIN User or remote access PIN

Function	0 (Opens the Help menu.)
	1 (Opens the main menu.)
	2 (Plays back new messages.)
	3 (Plays back all of the messages.)
	4 (Jumps back to the previous message.)
	5 (Repeats the message currently being played back.)
	6 (Jumps to the next message.)
	7 (Deletes the message currently being played back.)
	8 (Ends playback.)
	* 10 (Disables the voice mailbox.)
	* 11 (Enables the voice mailbox.)
	* 20 (Enables recording and changing announcements.)
Additional information for * 20	The default announcement and your announcements are played, if present. The following functions are available whilst your own announcements are being played back:
	# (Selects the currently heard announcement as active announcement.)
	4 (Jumps to the previous announcement.)
	5 (Repeats the announcement you have just heard.)
	6 (Jumps to the next announcement.)

	7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)
Function	* 30 (Disables message forwarding.)
	* 31 (Enables message forwarding.)
	* 70 (Deletes all messages and voice memos.)
	* 99 (Records a voice memo.)
	Hang-up (Ends remote access.)
	Additional function while a message is being replayed:
	9 (Announces the phone number of the caller – provided it has been transferred.)
	Further options after announcement of the phone number:
	# (Initiates a callback.)
	* (Back to the message playback.)
Additional information	After having dialled the internal number, press * while the announcement is being played.
	After having pressed #, wait until the announcement has ended

Further help under Voice mailbox

Remote Access (on an External Phone If Readiness of the Voice Mailbox Is Disabled)

Voice mailbox number > * > PIN > # > Function

Voice mailbox number	External number of the voice mailbox or
	External number of the phone which is owner/ user of the voice mailbox
PIN	User or remote access PIN
Function	0 (Opens the Help menu.)
	1 (Opens the main menu.)
	2 (Plays back new messages.)
	3 (Plays back all of the messages.)
	4 (Jumps back to the previous message.)
	5 (Repeats the message currently being played back.)
	6 (Jumps to the next message.)
	7 (Deletes the message currently being played back.)
	8 (Ends playback.)
	* 10 (Disables the voice mailbox.)
	* 11 (Enables the voice mailbox.)

	* 20 (Enables recording and changing announcements.)
Additional information for * 20	The default announcement and your announcements are played, if present. The following functions are available whilst your own announcements are being played back:
	# (Selects the currently heard announcement as active announcement.)
	4 (Jumps to the previous announcement.)
	5 (Repeats the announcement you have just heard.)
	6 (Jumps to the next announcement.)
	7 (Selects the currently heard or free announcement for changing or recording. Press # to save your announcement or * to cancel the function.)
Function	* 30 (Disables message forwarding.)
	* 31 (Enables message forwarding.)
	* 70 (Deletes all messages and voice memos.)
	* 99 (Records a voice memo.)
	Hang-up (Ends remote access.)
	Additional function while a message is being replayed:
	9 (Announces the phone number of the caller – provided it has been transferred.)

Voice mailbox

Further options after announcement of the phone number:

(Initiates a callback.)

* (Back to the message playback.)

Additional information

After having dialled the internal number, press * while the announcement is being played. The voice mailbox accepts the call after up to 55 seconds.

After having pressed #, wait until the announcement has ended



Further help under Voice mailbox

Allowlist (incoming)

Enabling allowlist (Incoming) for users (on an internal phone)

8 > PIN > * 22 1

PIN User PIN

Further help under Block- and allowlist

Disabling allowlist (Incoming) for users (on an internal phone)

8 > PIN > * 22 0

PIN User PIN

Further help under Block- and allowlist

Hardware

Information about the section Module LEDs
Button

Information about the section

Here you can find information about various hardware platforms of the COMtrexx.



For further information, take a look at the Instruction of your COMtrexx variation.

Module

The following modules are available for extending the COMtrexx Business:

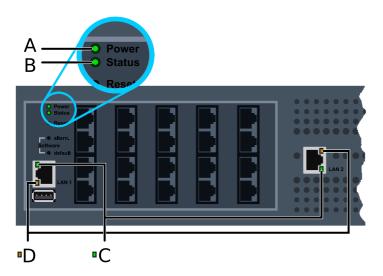
- NET module: By upgrading with a NET module, there are now two networks with different IP address ranges available.
 - Further help under Network interface (second)
- 4FXS Module: By upgrading with a 4FXS module, there are now four internal ports for analogue devices (telephones, fax machines, door stations, announcement outputs) available.
 - Further help under <u>Fax</u>
 - Further help under **Door functions**
 - Further help under <u>Loudspeaker Audio Out</u>
 - Important: Installed and configured 4FXS modules reduce the available VoIP operation modes and therefore also the maximum number of available VoIP channels.
 - Further help under VolP

Configuration

- · Configuring installed modules.
 - Hardware

LEDs

The COMtrexx Business offers the following LEDs for status display:



- A LED Power
- B LED Status
- C LED LAN link
- D LED LAN activity

User guidance

Status LED:	User prompt:
Flashes red (50/50 ms)	Pressing the Software default button is detected.
Flashes red (0,5/0,5 s)	Button Software default was pressed long enough.

Status LED:	User prompt:
Flashes orange (50/50 ms)	Pressing the Software altern. button is detected.
Flashes orange (0,5/0,5 s)	Button Software altern. was pressed long enough.

Status of the PBX

Status of the PBX		
Status LED:	Power LED:	PBX status (after restart):
Lit in red	Lit in red	The kernel is being loaded from the boot partition.
Lit in orange	Lit in red	The kernel is being initialised and the partition is checked.
Lit in green	Lit in red	The system is being initialised.
Lit in green	Lit in orange	The kernel modules are being loaded.
Lit in green	Flashes orange	The database is being initialised/converted.
Lit in green	Lit in green	The PBX is ready for operation.
Status LED:	Power LED:	PBX status (in the case of an error):
Flashes red (1/1 s)	Lit in red	No valid boot partition was found. System is paused.

Status LED:	Power LED:	PBX status (in the case of an error):
Flashes orange (1/1 s)	Lit in red	No consistent kernel data has been found on either boot partition. System is paused.

Note: If the Power LED remains constantly lit in red even after a restart, an error has occured. Please contact your dealer or the manufacturer directly.

Status LED:	Power LED:	PBX status (during powering down):
Lit in green	Flashes green	The delayed powering down has been enabled and the PBX waits for the termination of the last call.
Lit in green	Lit in red	The PBX is powered down completely.

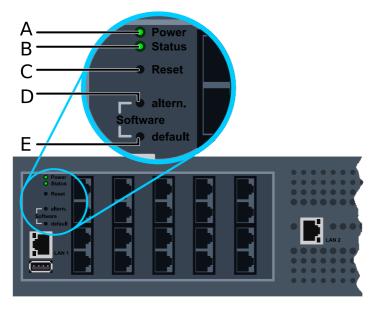
Ethernet interface

LAN-Link LED:	Status of the Ethernet Interface:
Off	No Ethernet connection to a computer/network exists.
Lit in green	An Ethernet connection to a computer/ network exists (link).

LAN-Activity LED:	Transfer Activities on the Ethernet Interface:
Off	No transfer activity on the Ethernet interface.
Flashes orange	Packets are being transmitted.

Button

The COMtrexx Business offers the following buttons for proceeding service functions:



- A LED Power
- B LED Status
- C Button Reset
- D Button Software altern.
- E Button Software default

Restart

To initiate a restart, proceed as follows:

▲ Caution: This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!

Press the Reset

The LEDs indicate the following states:

- The **Power** LED lights up/flashes in red and orange for up to two minutes
- If the **Power** LED lights up in green again, the restart is complete and the PBX is ready for operation.
 - Note: If the Power LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.
- Further help under Restart

Restoring firmware

To restore the previous firmware (e.g. in case of problems after an update), proceed as follows:

- **Important:** First read the steps to be taken.
- ▲ Caution: Afterwards a restart is carried out automatically. This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!
- 1. Successively press the buttons **Reset** and **Software altern.** and keep the button Software altern. pressed.

The LEDs indicate the following states:

- The **Status** LED starts to quickly blink in orange.
- After a short time, the **Status** LED will slow down.

2. Let go of the Software altern. button.

The LEDs indicate the following states:

- The **Power** LED lights up/flashes in red and orange for up to two minutes.
- If the Power LED lights up in green again, the restart is complete and the PBX is ready for operation.
 - Note: If the Power LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.
- Further help under Firmware Update

Factory settings

To restore the factory settings, proceed as follows:

- Important: First read the steps to be taken.
- ▲ Caution: Afterwards a restart is carried out automatically. This interrupts all the connections with/to the web server, all the calls and the data services. Unsaved data is lost!
- 1. Successively press the buttons **Reset** and **Software default** and keep the button **Software default** pressed.

The LEDs indicate the following states:

- The **Status** LED starts to quickly blink in red.
- After a short time, the Status LED will slow down.
- 2. Let go of the Software default button.

The LEDs indicate the following states:

Button

- The **Power** LED lights up/flashes in red and orange for up to two minutes.
- If the Power LED lights up in green again, the restart is complete and the PBX is ready for operation.
 - Note: If the Power LED remains continuously lit in red, an error has occurred. Please contact your dealer or the manufacturer directly.
 - Note: If you let go of the Software default button too late (the Status LED flashes no more), the PBX restarts without changes.

Glossary

Glossary

Technical Terms and Functions Abbreviations

Technical Terms and Functions

Account

APIPA

Call Deflection

Certificate

Certificate Authority

Client

CLIP no screening

Codec

DDI

DECT

DHCP

DiffServ

DNS

Domain

Driver

DSL

DTMF

DynDNS

Echo Cancellation

End device

Ethernet

Exchange line

Private branch exchange

Firewall

Firmware

Flash

GAP

Gateway

Technical Terms and Functions

Glossary HOST HTTP proxy **HTTPS Hyperlink IAD IP** address IPv4 address IPv6 address Jitter buffer LAN **LDAP LED MAC** address **MSN**

NAT

Network

NTP

Outbound proxy

PD

PIN

PoE

Port

Prefix

Proxy server

PTMP connection

PBX connection

Central office

QoS

QR code

Technical Terms and Functions

Registrar External private branch exchange Router **Routing** Server **SIP SIP** port **SIP-DDI (SIP Trunking) SMTP Softphone STUN** Subnet mask **Switch Syslog server** System telephone **TAPI TCP Template T-Net Functions UDP URL VoIP VoIP** account **VoIP** provider

Voucher VPN WAN

Account

Authorisation to access a VoIP provider or a VoIP PBX. Users must identify themselves by logging in with a user name and a password.

APIPA

Automatic Private IP Addressing - allows network interfaces to be configured automatically. A device that supports APIPA allocates itself a random IP address from the range 169.254.1.0 to 169.254.254.255 or has a permanent IP address from the APIPA address range.

Call Deflection

Designation for call forwarding during the ringing period. If Call Deflection has been activated, the called person can deflect the call on a case-to-case basis while the phone is still ringing.

Certificate

A (digital) certificate is similar to a digital passport that serves to identify the owner of the certificate. Its authenticity can be verified by cryptographic procedures. The certificate thus confirms the identity of a person or a company by means of an asymmetric key, thus ensuring that the person or company really is the person/company it claims to be. Certificates are issued by an officicial institution, the Certification Authority (CA).

Certificate Authority

A certificate authority is an organisation that issues digital certificates. A digital certificate serves to assign a specific public key to a person or organisation. The certificate authority certifies this assignment by adding its own digital signature.

Client

Client

Piece of computer hardware or software that accesses a service made available by a server.

CLIP no screening

Phone service for outgoing calls that allows the party called to see a customer-defined phone number instead of the phone number assigned by the network provider.

In this context, no screening means that the customer-defined number is not checked by the public exchange against the assigned numbers. It may contain any phone number you define (e.g. a service phone number). Codec

Codec

Method to encode (digitise) analogue voice data for transmission and decode it again to convert it back into voice. There are various codecs that feature different voice data compression rates thereby require different band widths for data transmission. The quality of VoIP calls depends on the codec used.

DDI

Direct Dial In - extension number for dialling directly to a specific user on a phone system. Example: In the number 0 53 06 / 92 00 - 700, 700 is the Direct Dial In number (DDI) for the Auerswald info-line.

DECT

Digital Enhanced Cordless Telecommunications (previously Digital European Cordless Telephony) - a standard for digital cordless phones and data transmission (also see <u>GAP</u>). Auerswald DECT systems provide encrypted transmission between mobile handset and base station.

DHCP

DHCP

The IP address is temporarily allocated for a certain amount of time. If the address is no longer required by the client, the server has access to it again, and can allocate it to another client.

DiffServ

Differentiated Services Flag - flag 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.

DNS

Domain Name Service - needed to translate Internet addresses. The name of a computer on the Internet (for example, Auerswald) is assigned to the corresponding IP address. This service is provided by DNS servers at the various Internet service providers or by upper domain servers.

Domain

Globally unique name of a website, consisting of third-level domain (e.g. the service name -www), second level domain (e.g. auerswald) and top-level domain (e.g. the country code de). The domain is part of the URL.

Driver

Driver

Programme or software module that controls the interaction between an operating system or programme and the interfaces to connected devices (keyboard, printer, monitor, etc.) or virtual devices.

DSL

DSL

Digital Subscriber Line - Internet access technology using a digital transmission standard that provides wide bandwidths for the transmission of data via telecommunication networks.

DTMF

DTMF

Dual Tone Multi Frequency dialling - dialling method that transmits the dialled number to the exchange as two superimposed sinusoidal signals.

DynDNS

DynDNS services make it possible to access an Internet connection despite a dynamic IP address. To do this, the service provider assigns the changing IP address of an Internet connection to a permanent domain name (e.g. YourName.DynDNS.org).

Echo Cancellation

Local echos and reverberation effects are compensated using echo cancellation. For example, an echo is generated on the receiver of a phone; the voice of the caller emanates from the loudspeakers and reaches the microphone where it is then transmitted back.

End device

Device that can be operated on a communication network or on a PBX, e.g. phone, fax machine, answering machine etc.

Ethernet

Network system with a speed of 10/100/1000 Mbit/s that was developed by the companies INTEL, DEC and Xerox.

Exchange line

General term for a connection that links the PBX to the public switched phone network. The exchange line joins the provider's local exchange and the phone connection.

Private branch exchange

Internal connections of a PBX together with the devices, e. g. phones and telefaxes, connected to them.

Firewall

Firewall

Network security component that uses a set of security rules to protect a computer network or an individual computer against unauthorised access to or from the network.

Firmware

Operating software on the phone or PBX that is stored in the device itself. The device cannot function without firmware.

Flash

Flash

Data storage device (e.g. USB memory stick) for small-footprint non-volatile data storage without a permanent power supply.

GAP

Generic Access Profile - transmission protocol for cordless phones.

Describes the basic functions for the communciation of DECT devices from different manufacturers among themselves.

Gateway

PC or router that acts as an intermediary between two networks. The Internet service provider is the gateway for direct Internet dial-up connections. If you use a router, this is the gateway in a local network.

Glossary

Glossary

HOST

Component of a data processing system that manages larger application programs and data volumes and makes them available to lower-order servers and clients.

HTTP proxy

Proxy for the HTTP protocol, over which Internet sites are called or files downloaded.

HTTPS

Hypertext Transfer Protocol Secure - communication protocol for tap-proof data transfer with encryption within the World Wide Web.

Hyperlink

Hyperlink

Cross reference, allowing to jump to a defined location within or a different document (e.g. inside an Internet document).

IAD

The IAD (Integrated Access Device) is a device for realising a connection (such as a DSL router) that usually is provided by the Internet provider for external VoIP telephony and Internet access.

IP address

Unique numerical address within a TCP/IP network that is assigned to one device and ensures that data packages reach the correct recipient.

IPv4 address

Certain ranges are reserved for operating local networks:

- Class A: 10.x.x.x (for networks with up to 16.5 million PCs)
- Class B: 172.16.x.x to 172.31.x.x (for networks with up to 65534 PCs)
- Class C: 192.168.0.x to 192.168.255.x (for networks with up to 254 PCs)

These addresses have no validity on the Internet. This means that data packets with this kind of sender or recipient address cannot be sent over the Internet. However, they can be used in local networks without restriction. The benefit of this is that if data from a local network configured in this way comes in contact with the Internet, none of the data on the computers in the local network can leak out or be accessed externally.

IPv6 address

IPv4 addresses comprise eight blocks with four digits or characters each that are separated by a colon, for example, 3001:00FF:ABC0:0EAC:0001:0000:0000:000F, or, abbreviated, 3001:FF:ABC0:EAC:1::F.

Thanks to the greater coding, IPv6 offers a greater number of available IP addresses than version 4 of the Internet protocol (IPv4).

Jitter buffer

Buffers RTP packages, to compensate irregular transfer rates of the Internet.

A high jitter buffer reduces short interruptions during the call. In return the delay of the RTP packages increases.

LAN

Local Area Network - connects computers over company or home networks.

LDAP

Lightweight Directory Access Protocol) - enables you to query the data in a directory service (a hierarchical database) over the network.

LED

Light emitting diode - An electronic device similar to a lamp and used for displays (also multi-coloured).

MAC address

Media Access Control address - specific, unique hardware address specified for each individual single network card used in the whole world. This means that the network card is unique and can be controlled without any problems and detected by the network. The IP addresses on the TCP/IP network are always mapped to the respective MAC address.

MAC addresses are represented in a byte-by-byte format, with the individual bytes being separated by colons or hyphens, e.g. 00:09:52:04:4B:13.

MSN

Multiple Subscriber Numbers - multiple phone numbers where an PTMP connection can be reached. The MSN is the part of the phone number without a local prefix.

NAT

Network Address Translation - permits the mapping of a specific IP address used within a network into another IP address used by another network. This function is e.g. performed by a router which connects a local area network to the Internet.

Network

System of more than one computer and other communication devices This enables multiple users to access common resources such as files, printers, etc.

NTP

Network Time Protocol - standard for time synchronization of devices in packet based communication networks. NTP uses the connectionless transport protocol UDP. It allows a reliable time display in networks with packet delay variation.

Outbound proxy

Intermediate server that processes all VoIP requests and connections going to the provider (except registration).

PD

PD

Pulse dialling - an older dialling method that transmits the dialled number to the exchange by interrupting the current in the A-wire of the phone connection.

PIN

Personal Identification Number - multi-digit secret code that has to be entered to get access to services or authorisations.

PoE

PoE

Power over Ethernet. Here, network compatible devices can be powered over the eight-wire ethernet cable. A seperate power cable is not necessary.

Port

Port

A single PC can simultaneously establish multiple connections and provide multiple services for other PCs. Ports are used to differentiate these connections from parallel connections.

Due to a common agreement, PCs usually provide their services on ports 1 to 1023. Outgoing connections are usually established starting at Port 1024. Most services use the standard port numbers (for example, web browsers use port 80).

Prefix

Prefix

Number range containing the prefix or all of the prefixes from the area that should be collected for the network (e.g. local phone network, regional network, international network).

Proxy server

Acts as the interface to the Internet or between two networks. A proxy server receives queries from a computer in an Intranet, for example. It then connects to the Internet (on behalf of the computer) to forward the queries and return the (if necessary, filtered) responses. To achieve this, the HTTP proxy uses the Internet's HTTP protocol.

PTMP connection

Point-to-Multipoint connection with a certain number of independent phone numbers (MSNs). They can be assigned to one or more devices, including PBXs.

PBX connection

PTP connection with Direct Dial In option. A provider allocates a PBX base number with additional Direct Dial In numbers (DDIs), e.g. 9200–0. PBXs are usually operated on a PTP connection.

Central office

Node point on the public telecommunication network.

QoS

Quality of service - service of the router or switch. This function provides constantly high speech quality by prioritising speech data (RTP data packages).

QR code

QR code

Quick Response Code - two-dimensional code for binary representation of data in a matrix of black and white squares: \$\mathbb{\mathbb{R}}\ext{0}\$.

Registrar

A registrar (domain name registar) is an organisation or a company that registers Internet domains.

External private branch exchange

Internal user (e.g. home office, outlying parts of the building, branch offices) connected to a central phone system via a public network (e.g. the Internet). The distance between the phone system and the external private branch exchange is not relevant.

Router

Router

Routers connect two separate networks. This means, for example, that you can connect the local network to another LAN or a WAN. When a PC wants to send a data packet, this packet must travel over a router. The router uses the IP address to detect the network it must send the data to.

In addition to connecting to networks, routers can also execute certain control functions, such as maintaining a simple firewall. In a home network, routers are usually used to connect the local network to the Internet and to enable simultaneous Internet access for multiple users.

Routing

Routing

Identification of paths for the flow of data within computer networks. Routing determines the complete path of a set of data through the network.

Server

Server

Computer or a software that fulfils different tasks as part of the network and provides, for example, other users (clients) in this network with certain information, data and services.

SIP

Session Initiation Protocol - network protocol which provides establishing a communication session between two or more users. SIP is only used to negotiate the communication conditions. Other protocols such as RTP are responsible for actual data transfer.

SIP port

Port of the local system used as the communication port for the SIP transfer.

SIP-DDI (SIP Trunking)

Some VoIP providers also support a VoIP PTP connection with a Direct Dial In option. The provider allocates a PBX base number with additional Direct Dial In numbers (DDI numbers), e.g. 9200–0.

SMTP

Simple Mail Transfer Protocol - network protocol used for the secure transfer of e-mails between the PBX and external mail servers. The PBX needs an external SMTP server (outgoing mail server) to send system e-mails or forward voice mail messages and fax messages.

Under **Administration > E-mail > Outgoing mail server** you can enter different outgoing mail servers. If no outgoing mail server has been entered, no e-mails can be sent.

Softphone

Softphone is short for software telephone. This is an app (also known as a soft client) for conducting VoIP calls via a mobile phone or computer.

STUN

Simple Transversal of UDP over NATs - used to determine the public IP address of an Internet connection. To do this, a STUN request is sent to a STUN server, which then shares its own IP address allocated by the VoIP provider with the Internet connection.

Subnet mask

The value of the subnet mask, as with IP addresses, is often given in decimal form (for example, 255.255.255.0 for IPv4 addresses).

Switch

Switch

Active network distributor that distributes data packages to a destination defined by an IP address among the different segments of a network.

Syslog server

Records protocols with control and management information about the PBX, which helps to identify errors and, where possible, their causes.

System telephone

Thanks to their integration into the system, system telefphones simplify the operation of the entire system and additionally offer numerous possibilities to use PBX functions and signalise their status visually and, in part, accustically.

TAPI

Phone Application Programming Interface - name of a standardised software interface for computer-assisted telephony (CTI). TAPI ensures that software and hardware from various manufacturers are compatible with each other.

TCP

Transmission Control Protocol - transport protocol that segments data into packets up to a specified size and reliably sends these individual data packets in the correct sequence to the recipient address. In this process, every data packet sent must be resent until it has been confirmed as arrived. To ensure this happens, a large amount of information is sent along with the actual payload data. Most Internet services are implemented with TCP, e.g. HTTP (WWW), SMTP/POPS (e-mail), etc.

Template

Template, providing a basic configuration for a PBX or a telephone.

A provisioning file for a telephone for example provides certain default settings that can later on be complemented by user-specific data.

A provider template sets a configuration for a PBX that makes the configuration with a certain provider easier and allows a preferably problem-free operation.

T-Net Functions

Functions available on the analogue T-Net in Germany. There are analogue comfort phones with pre-defined function keys or menus for T-Net functions such as call forwarding, toggling between conversations or conference calling.

UDP

User Datagram Protocol) - protocol used to send data packets over non-secure communication lines without a connection. This means that successful transmission is dependent on the application and is therefore not always guaranteed. UDP itself does not verify whether data has been transmitted successfully. When a UDP packet is sent, the sender cannot assume that the packet will indeed arrive at the recipient. This particular protocol needs only a small amount of additional information, and results in a better data throughput rate in a well-functioning network, e.g. on a LAN. UDP is used by DNS servers, for example.

URL

URL

Uniform Source Locator - complete address of a resource (e.g. a website), consisting of a scheme (e.g. http:// and a scheme-specific part (e.g. domain www.auerswald.de/ and a path en/products/pbx/home-office.html).

VoIP

VoIP

Voice over Internet - Internet telephony. Instead of an analogue or ISDN line, the Internet connection (for example, DSL) is used for telephoning. In this case, digital voice data is sent as IP packets from one telephone to another. This functions the same way as the transmission of a webpage over the Internet. The transmission quality and the reliability of Voice over IP depends to a great extent on the quality of the Internet connection used.

VoIP account

Account configured with a VoIP provider who provides the necessary access data required for VoIP calls. To do this, use your name and address data to register on the provider's website. After that, you will be assigned one or more telephone numbers that can be reached from the land line and the Internet, and also an account with a user name and password. The registered connection usually becomes active within a few minutes and can be used very soon after that.

VoIP provider

Internet service provider offering Internet telephony (VoIP, Voice over Internet Protocol).

Voucher

Voucher

Voucher for a function release for a specific device type.

VPN

VPN

Virtual Private Network - connects remote or neighbouring networks (LAN) to the internal network using a VPN tunnel. The tunnel functions basically as a channel in which embedded data is transported from one network to the other.

WAN

Wide Area Network - a public network, e.g. the Internet.

Abbreviations

AOCD	Advice Of Charge During Call	Charge information during and at the end of the call
AOCE	Advice Of Charge End of Call	Charge information at the end of the call
<u>APIPA</u>	Automatic Private IP Addressing	
CF		Call forwarding
BLF	Busy Lamp Field	Busy Lamp Field
CAPI	Common ISDN Application Programming Interface	
CCBS	Completion of Calls to Busy Subscriber	Recall on busy
CCNR	Completion of Calls on No Reply	Recall on no reply
CD	Call Deflection	call forwarding via the called person
CD (PR)	Call Deflection (Partial Rerouting)	call forwarding of direct dial in numbers in the public exchange
CFB	Call Forwarding on Busy	Enable Call Forward Busy

CFNR	Call Forwarding on No Reply	Call forwarding on no reply
CFU	Call Forwarding Unconditional	Call forwarding immediately
CIDR	Classless Inter-Domain Routing	
CLIP	Calling Line Identification Presentation	Phone number display
CLIR	Calling Line Identification Restriction	Phone number is not displayed if that is appropriate
CNIP	Calling Name Identification Presentation	Calling Name Identification Presentation
CNIR	Calling Name Identification Restriction	Calling Name Identification Restriction
COLR	Connected Line Identification Restriction	Connected Line Identification Restriction
CoNDIS	Connection-Oriented Network Driver Interface Specification	
CSV	Comma-Separated Values	
СТІ	Computer Telephony Integration	Computer Telephony Integration
<u>DDI</u>	Direct Dial-In	Extension

DECT	Digital Enhanced Cordless Telecommunications	Digital Enhanced Cordless Telecommunications
dial-up connection		dial-up connection
DHCP	Dynamic Host Configuration Protocol	
<u>DiffServ</u>	Differentiated Services Flag	
DIP	Dual Inline Package	
DNS	Domain Name Service	
DSCP	Differentiated Services Code Point	
DSL	Digital Subscriber Line	Digital Subscriber Line
DSP	Digital Signal Processor	Digital Signal Processor
DTMF	Dual Tone Multi Frequency	Dual Tone Multi Frequency
<u>DynDNS</u>	Dynamic DNS	Dynamic DNS
ECM	Error Correction Mode	Error Correction Mode
Loudspeal system		Loudspeaker system
EMC		Electro-Magnetic Compatibility
GAP	Generic Access Profile	

GSM	Global System for Mobile communication	Global System for Mobile communication
HLC	High Layer Capability	
HTML	Hypertext Markup Language	
HTTP	Hypertext Transfer Protocol	Hypertext Transfer Protocol
HTTPS	Hypertext Transfer Protocol Secure	Hypertext Transfer Protocol Secure
IAD	Integrated Access Device	
IP	Internet Protocol	Internet Protocol
IPUI	International Portable User Identity	International Portable User Identity
<u>PD</u>		PulseDialling
LAN	Local Area Network	Lightweight Directory Access Protocol
LDAP	Lightweight Directory Access Protocol	Protocol for easy access to directory server
<u>LED</u>	Light-emitting diode	
MAC	Media Access Control	
MWI	Message Waiting Indication	
<u>DTMF</u>		Dual TMrequency

MSN	Multiple Subscriber Number	
NAT	Network Address Translation	
NF		Low frequency
<u>NTP</u>	Network Time Protocol	
PCL	Printer Command Language	
PIN	Personal Identification Number	
PMP	Protect Monitoring Point	
<u>PoE</u>	Power over Ethernet	
PPP	Point-to-Point Protocol	
PTMP	Point-to-Multipoint	PTMP connection
PTP	Point-to-Point	PBX connection
<u>QoS</u>	Quality of service	
RJ	Registered Jack	Standardised socket
RSS	Really Simple Syndication	
RTP	Real-Time Transport Protocol	
SD	Secure Digital	
SDHC	Secure Digital High Capacity	

Abbreviations

SFTP	Secure File Transfer Protocol	
SIP	Session Initiation Protocol	
SIPS	Session Initiation Protocol Secure	
SMTP	Simple Mail Transfer Protocol	
SRTP	Secure Real-Time Transport Protocol	
STUN	Simple Transversal of UDP over NATs	
SW	Software	
TAPI	Phone Application Programming Interface	
TCP	Transmission Control Protocol	
TSM		door terminal module
UAE		
<u>UDP</u>	User Datagram Protocol	
CF		Call forwarding

Abbreviations

<u>URL</u>	Uniform Resource Locator	Uniform Resource Locator
USB	Universal Serial Bus	
UPS		Uninterrupted Power Supply
UTC	Coordinated Universal Time	
VMF	Voice Mail and Fax	
VoIP	Voice over Internet Protocol	
<u>VPN</u>	Virtual Private Network	
WAN	Wide Area Network	
XML	Extensible Markup Language	