MIVOICE OFFICE 400

MITEL 470

AS OF VERSION R6.0

SYSTEM MANUAL

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NOTICE

TRADEMARKS

Mitel 470

syd-0585/2.0 – 09.2018

Content

1 Product and Safety Information

Here you will find information relating to safety, data protection and legal matters besides product and documentation information.

Please read through the product and safety information carefully.

1. 1 About MiVoice Office 400

Purpose and function

MiVoice Office 400 is an open, modular and comprehensive communication solution for the business sector with several communication servers of different performance and expansion capacity, an extensive telephone portfolio and a multitude of expansions. They include an application server for unified communications and multimedia services, an FMC controller for mobile phone integration, an open interface for application developers, and a multitude of expansion cards and modules.

The business communication solution with all its components was developed to cover in full the communication requirements of businesses and organisations, in a way that is both user- and maintenance-friendly. The individual products and components are coordinated and must not be used for other purposes or replaced by third-party products or components (unless it is to connect other approved networks, applications and terminals to the interfaces certified specially for that purpose).

User groups

The design of the phones, softphones and PC applications of the MiVoice Office 400 communication solution is particularly user-friendly, which means they can be operated by all end users without specific product training.

The phones and PC applications for professional applications, such as the operator console or call centre applications require training of the personnel.

Specialist knowledge of IT and telephony is assumed for the planning, installation, configuration, commissioning and maintenance. Regular attendance at product training courses is strongly recommended.

User information

MiVoice Office 400 products are supplied with the necessary safety/legal information and user documents. All user documents such as user guides and system manuals are available for download from the MiVoice Office 400 document portal as individual documents or as documentation sets. Some user documents are accessible only via a partner login.

It is your responsibility as a specialist retailer to keep up to date with the scope of functions, the proper use and the operation of the MiVoice Office 400 communication solution and to inform and instruct your customers about all the user-related aspects of the installed system:

- Please make sure you have all the user documents required to install, configure and commission a MiVoice Office 400 communication system and to operate it efficiently and correctly.
- Make sure that the versions of the user documents comply with the software level of the MiVoice Office 400 products used and that you have the latest editions.
- Always read the user documents first before you install, configure and put a MiVoice Office 400 communication system into operation.
- Ensure that all end users have access to the user guides.

Download the MiVoice Office 400 documents from the internet:
http://www.mitel.com/docfinder or from http://edocs.mitel.com

1. 2 Safety Information

Reference to hazards

Hazard warnings are affixed whenever there is a risk that improper handling may put people at risk or cause damage to the MiVoice Office 400 product. Please take note of these warnings and follow them at all times. Please also take note in particular of hazard warnings contained in the user information.

⚠ **DANGER**!
Danger indicates an imminently hazardous situation which, if not avoided, will result in death or serious injury.

⚠ **WARNING**!
Warning indicates a potentially hazardous situation which, if not avoided, could result in death or serious injury.

⚠ **CAUTION**!
Caution indicates a potentially hazardous situation which, if not avoided, may result in minor or moderate injury and/or damage to the equipment or property.

These symbols may appear on the product:

The lightning flash with arrowhead symbol, within an equilateral triangle, is intended to alert the user to the presence of uninsulated dangerous voltage within the product's enclosure that may be of sufficient magnitude to constitute a risk of electric shock.

The exclamation point within an equilateral triangle is intended to alert the user to the presence of important operating and maintenance (servicing) instructions in the literature accompanying the product

Indicates ESD components. Failure to observe information identified in this way can lead to damage caused by electrostatic discharge.

The ground symbol within a circle identifies the product to be connected to an external conductor. Connect this product to earth ground before you make any other connections to the equipment.

Operating safety

MiVoice Office 400 communication servers are operated on 115/230 VAC mains power. Communication servers and all their components (e.g. telephones) will not operate when mains power fails. Interruptions in the power supply will cause the entire system to restart. A UPS system has to be connected up-circuit to ensure an uninterruptible power source. Up to a specific performance limit a Mitel 470 communication server can also be powered redundantly using an auxiliary power supply. For more information please refer to your communication server's system manual.

When the communication server is started for the first time, all the configuration data is reset. You are advised to backup your configuration data on a regular basis as well as before and after any changes.

Installation and operating instructions

Before you begin with the installation of the MiVoice Office 400 communication server:

- Check that the delivery is complete and undamaged. Notify your supplier immediately of any defects; do not install or put into operation any components that may be defective.
- Check that you have all the relevant user documents at your disposal.
- Configure this product with only the assemblies specified and in the locations stated in the user documentation.
- During the installation follow the installation instructions for your MiVoice Office 400 product in the sequence that is given and observe to the safety warnings they contain.

⚠ **CAUTION!**
Failure to follow all instructions may result in improper equipment operation and/or risk of electrical shock.

- Install all wiring according to local, state, and federal electrical code requirements.
- Do not connect telecommunications cabling to the system, service the system, or operate the system with the grounding conductor disconnected.
- Ensure the AC receptacle is installed near the equipment and easily accessible.
- Use only Mitel approved power adapters.

Any servicing, expansion or repair work is to be carried out only by trained technical personnel with the appropriate qualifications.

1. 3 Data protection

Protection of user data

During operation the communication system records and stores user data (e.g. call data, contacts, voice messages, etc.). Protect this data from unauthorised access by using restrictive access control:

- For remote management use SRM (Secure IP Remote Management) or set up the IP network in such a way that from the outside only authorised persons have access to the IP addresses of the MiVoice Office 400 products.
- Restrict the number of user accounts to the minimum necessary and assign to the user accounts only those authorisation profiles that are actually required.
- Instruct system assistants to open the remote maintenance access to the communication server only for the amount of time needed for access.
- Instruct users with access rights to change their passwords on a regular basis and keep them under lock and key.

Protection against listening in and recording

The MiVoice Office 400 communication solution comprises features which allow calls to be monitored or recorded without the call parties noticing. Inform your customers that these features can only be used in compliance with national data protection provisions.

Unencrypted phone calls made on the IP network can be recorded and played back by anyone with the right resources:

- Use encrypted voice transmission (Secure VoIP) whenever possible.
- For WAN links used for transmitting calls from IP or SIP phones, use as a matter of preference either the customer's own dedicated leased lines or with VPN encrypted connection paths.

1. 4 About this document

This document contains information on the expansion stages, system capacity, installation, configuration, running and maintenance as well as the technical data of the MiVoice Office 400 communication servers. The system functions and features, the DECT planning and the possibilities for networking several systems into a private network (PISN) or an Mitel Advanced Intelligent Network (AIN) are not part of this Manual; they are described in separate documents.

Note
In this document, it is presumed, that the Mitel SMB Controller is loaded with a MiVoice Office 400 application software. This assumption is always valid, even the expression Mitel SMB Controller, SMBC or communication server is used.

The expansion possibilities for the Mitel 470 communication server include an applications server for unified communications and multimedia services, an FMC Controller for integrating mobile/external phones, an open interface for application developers and a multitude of expansion cards and modules.

The document is intended for planners, installers and system managers of phone equipment. Basic knowledge of phones, especially ISDN and IP technology, is required to understand the content.

The system manual is available in Acrobat Reader format and can be printed out if necessary. Navigation in PDF format is based on the bookmarks, table of contents, cross references and index. All these navigation aids are linked, i.e. a mouse click takes you directly to the corresponding places in the Manual. We have also ensured that the page numbering in the PDF navigation corresponds to the page numbering of the Manual, making it much easier to jump to a particular page.

Referenced menu entries and parameters appearing on terminal displays or on the user interfaces of the configuration tools are *highlighted* in italics and in colour for a clearer orientation.

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  https://pbxweb.aastra.com/doc_finder/DocFinder/syd-0585_en.pdf?get&DNR=syd-0585

General Considerations

Special symbols for additional information and document references.

Note
Failure to observe information identified in this way can lead to equipment faults or malfunctions or affect the performance of the system.

Tip
Additional information on the handling or alternative operation of equipment.

See also
Reference to other chapters within the document or to other documents.

Mitel Advanced Intelligent Network
Particularities that have to be observed in an AIN.

References to the MiVoice Office 400 configuration tool WebAdmin

If an equals sign is entered in the WebAdmin search window, followed by a two-digit navigation code, the view assigned to the code is directly displayed.
Example: *Licence overview* (*=q9*) view
The corresponding navigation code is available on the help page of a view.

2 System Overview

This chapter provides a brief overview of the Mitel 470 communication server with its positioning within the MiVoice Office 400 series and the networking possibilities. It also features the system phones, the applications and the application interfaces. If you are setting up an communication system for the first time, it may be useful to set up a test system step by step on site. At the end of the chapter you find a useful getting started guide for this purpose.

2. 1 Introduction

MiVoice Office 400 is a family of IP-based communications servers for professional use in companies and organizations operating as small and medium-sized businesses in all industries. The family consists of four systems with different expansion capacities. The systems can be expanded using cards, modules and licences, and adapted to the specific requirements of companies.

The family covers the growing demand for solutions in the area of unified communications, multimedia and enhanced mobile services. It is an open system that supports global standards and is therefore easily integrated into any existing infrastructure.

With its wide range of networking capabilities the system is particularly well suited for companies that operate in several locations. Coverage can even be extended to the smallest branch offices at low cost.

MiVoice Office 400 communication systems handle “Voice over IP” technology with all its benefits. What’s more, the systems operate just as easily with traditional digital or analogue phones and public networks.

With the integrated Media Gateways any hybrid forms of an IP-based and digital or analogue communication environment are also possible. This enables customers to make the switch from traditional telephony to IP-based multimedia communication either in just one step or, gradually, in several stages.

2. 2 Communication server

Mitel 470 is a powerful communication server in the MiVoice Office 400 family. It is designed for installation in a 19” rack, but can also be set up on a flat surface.

With the exception of the power supply and earthing, all the connections and control elements are accessible from the front. The communications server does not have to be removed from the rack when expanding the system with interface cards, modules or an application card. Fig. 1 shows a Mitel 470 fitted with an application card and a number of interface cards.

Fig. 1 Mitel 470 with application card and a number of interface cards

The Mitel 470 communications server ships with a plug-in processor card (call Manager card) with colour display, 4 analogue terminal interfaces and 3 Gbit-LAN connections. A second CPU card (applications card) can be fitted as an option. It contains the pre-installed applications server for unified communications and multimedia services.

2. 2. 1 Positioning

Applications range from small businesses or branches to large companies at one or more locations. Up to 600 users can be operated on the Mitel 470 communication server. One licence is required for each user.

The diagram below shows the MiVoice Office 400 communication servers with their expansion capacity for users with SIP/IP phones and TDM extensions (FXS, DSI, BRI-S).

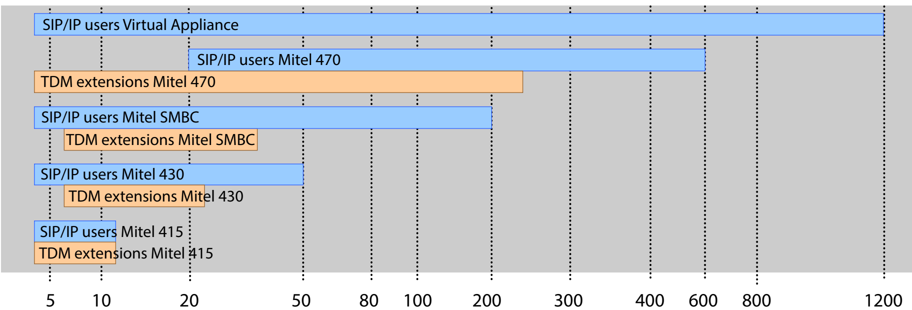


Fig. 2 Max. expansion capacity for users with SIP/IP phones and TDM extensions (FXS, DSI, BRI-S)

2. 3 Networking Possibilities

MiVoice Office 400 communication servers at different company locations, even beyond national borders, can be linked together to form an enterprise-wide private communication network with a common numbering plan. The following networking types are possible:

Mitel Advanced Intelligent Network (AIN)

In an AIN several communication servers of the MiVoice Office 400 series can be connected up to form a homogeneous communication system. The single systems are connected with one another via the IP network, thereby forming the nodes of the overall AIN system One node acts as the Master and controls the other (satellite) nodes. All the features are then available at all the nodes.

No call charges are incurred as the internal voice traffic between locations is routed via the system's own data network. All the AIN nodes are configured and set up centrally via the Master.

If a node is isolated from the rest of the AIN by an interruption in the IP connection, it restarts with an emergency configuration after a set amount of time. The connections are then routed to the public network via local links, for example with ISDN or SIP connections, until contact with the AIN is restored.

For the Virtual Appliance communication server, AIN networking (Virtual Appliance as master) with at least one satellite is mandatory.

SIP networking

Networking based on the open global SIP protocol is the universal way of connecting several systems with one another via the private data network or the internet. MiVoice Office 400 communication platforms can be used to network up to 100 other Mitel systems or SIP-compatible third-party systems. All the main telephony features such as call number and name display, enquiry call, hold, brokering, call transfer and conference circuits are supported. The transmission of DTMF signals and the T.38 protocol for Fax over IP between the nodes is also possible.

Virtual and leased-line networking via BRI/PRI interfaces

With this type of connection the nodes are connected via basic rate interfaces (BRI) or primary rate interfaces (PRI).

With virtual networking all the nodes are connected via the public ISDN network. This type of networking is particularly well suited for geographically dispersed locations which have such a low volume of calls between locations that leased lines or setting up a private data network are not worthwhile. The range of services available in a virtual network depends on the range of services offered by the network provider. The DSS1 ISDN protocol is the main protocol used.

With leased line networking the nodes are connected via dedicated or leased lines. One advantage of leased line networking are the fixed costs, regardless of the number of call connections. The most common protocol used is QSIG/PSS1, which supports several more features than the DSS1 protocol.

Virtual and leased-line networking can also be used in combination. Mitel systems are well as third-party systems can be used.

2. 4 Mitel system phones and clients

Mitel system phones stand out by virtue of their high level of user convenience and their attractive design. The broad range of products ensures there is a suitable model for every use.

Tab. 1 Mitel 6900 SIP series SIP phones

Product	Principal common features	Additional model-specific features
 Mitel 6920 SIP Phone  Mitel 6930 SIP Phone  Mitel 6940 SIP Phone	• User-friendly registration, configuration and operation of system features through MiVoice Office 400 integration. • XML browser compatible • Automatic update of the terminal software • Web-user interface • Integrated 1 Gbit Ethernet switch for connecting a PC • Hearing Aid Compatible (HAC) handset • Headset port convertible to DHSG/EHS capable headset port • Excellent voice quality due to Mitel Hi-Q™ wideband audio technology • Full-duplex hands-free operation (speakerphone) • Backlit display • Up to 3 expansion key modules can be connected • Three-party conference possible locally on the phone • Wall mounting possible • Power over Ethernet	Mitel 6920 SIP: • Corded speech optimized handset • MobileLink mobile device integration through optional USB Bluetooth Dongle • Magnetic keyboard connector • USB port 2.0 (100 mA) • Can be used as auxiliary reception phone (reduced functionality) in hospitality environments Mitel 6930 SIP: • Corded speech optimized handset • Support for optional cordless speech optimized handset • Magnetic keyboard connector • Can be used as auxiliary reception phone (reduced functionality) in hospitality environments Mitel 6930 SIP and Mitel 6940 SIP: • Cordless speech optimized handset • Mobile phone charging point • MobileLink mobile device integration • Bluetooth 4.1 interface • USB port 2.0 (500 mA) • Can be used as operator console • Mitel 6940 SIP • LCD touch display • Can be used as reception phone in hospitality environments General: • Additional model-specific features include the resolution, the display type and size, and the number of configurable or fixed function keys.

Tab. 2 Mitel 6800 SIP series SIP phones

Product	Principal common features	Additional model-specific features
 Mitel 6863 SIP Phone  Mitel 6865 SIP Phone  Mitel 6867 SIP Phone  Mitel 6869 SIP Phone  Mitel 6873 SIP Phone	• User-friendly registration, configuration and operation of system features through MiVoice Office 400 integration. • XML browser compatible • Automatic update of the terminal software • Web-user interface • Excellent voice quality due to Mitel Hi-Q™ wideband audio technology • Full-duplex hands-free operation (speakerphone) • Several configurable line keys • Three-party conference possible locally on the phone • Wall mounting possible • Power over Ethernet	Mitel 6863 SIP: • Integrated 10/100 Mbit Ethernet switch for connecting a PC Mitel 6865 SIP, Mitel 6867 SIP, Mitel 6869 SIP and Mitel 6873 SIP: • Integrated 1 Gbit Ethernet switch for connecting a PC • Backlit display • Expansion key modules can be connected • Headset socket (DHSG standard) Mitel 6867 SIP and Mitel 6869 SIP: • Magnetic keyboard connector • Can be used as auxiliary reception phone (reduced functionality) in hospitality environments Mitel 6867 SIP, Mitel 6869 SIP and Mitel 6873 SIP: • USB Interface • Replaceable keyboard covers Mitel 6869 SIP and Mitel 6873 SIP: • Can be used as operator console Mitel 6873 SIP: • Bluetooth interface • Can be used as reception phone in hospitality environments • LCD touch display General: • Additional model-specific features include the resolution, the display type and size, and the number of configurable or fixed function keys.

Note:
The phones of the Mitel 6700 SIP series (Mitel 6730 SIP, Mitel 6731 SIP, Mitel 6735 SIP , Mitel 6737 SIP , Mitel 6739 SIP, Mitel 6753 SIP, Mitel 6755 SIP and Mitel 6757 SIP) are supported as before (not all system features can be used).

Tab. 3 SIP Multimedia Terminal Mitel BluStar 8000i

Product	Main features
 Mitel BluStar 8000i	• Intelligent multimedia terminal with intuitive operation • Video conferencing solution, collaboration tool and application platform in one. • XML browser compatible • Bluetooth interface • Can be connected to a laptop • HD video camera with 30 frames per second • Three loudspeakers for voice transmission in HD audio quality • Four microphones to eliminate unwanted background noise • 13 inch colour touch-screen display • Biometric fingerprint reader • Desktop sharing • SIP-based

Tab. 4 IP system phones (softphones) and clients

Product	Main features
 Mitel BluStar for PC	• Autonomous and powerful SIP-based BluStar PC phone with video functionality • Can be used with headset or handset via PC audio interface, USB or Bluetooth • Graphical user interface with mouse and keyboard operation • User-friendly contact search • HD audio and HD video calls • Outlook integration • Link to e-mail client • Click to Call • Connection to an MS Lync server or an IBM Sametime server
 MiVoice 2380 Softphone	• Autonomous and powerful, IP-based PC system phone with intuitive user interface • Can be used with headset or handset via PC audio interface, USB or Bluetooth • Graphical user interface with mouse and keyboard operation • Displayable expansion keypad for team keys, functions and phone numbers • Displayable keypad • Ring tones expandable using .mp3, .mid and .wav files • Call contacts directly from Outlook • All the system features can be used
 MiVoice 1560 PC Operator	• OIP client application for a professional PC operator console • Can be used purely as an IP softphone (MiVoice 1560) or together with a system phone (MiVoice 1560) • Graphical user interface with mouse and keyboard operation • Can be used in an AIN as a network-wide PC operator console • Call management with internal and external queues • Presence indicator, presence profiles, phone book and journal • Operator groups and agent control • Line keys and calendar functions • Possibility of synchronisation with a Microsoft Exchange server • All the system features can be used

Product	Main features
 Mitel Office Suite	• OIP client application for PC-based call management • Used in conjunction with a system phone • Graphical user interface with mouse and keyboard operation • Configuration of the coupled system phone • Call manager with extensive functions and options • Presence indicator of other users • Configurable presence profiles • Phone book with address books and personal contacts • Journal with call lists, text messages and notes • Workgroups (agent control) • Possibility of synchronisation with a Microsoft Exchange server • Possibility of displaying various additional windows • All the system features can be used
 Mitel Mobile Client (MMC)	• FMC client for mobile phones (runs on various operating systems) • Integrates the mobile phone into the Mitel communication system • User is always reachable under the same call number (One Number concept) • Various telephone functions can be menu-operated both in the idle state and during a call • Other system features can be used via function codes • With MMC Controller handover is possible between internal WLAN and mobile radio network.

Tab. 5 MiVoice 5300 IP series IP system phones (hardphones)

Product	Principal common features	Additional model-specific features
 MiVoice 5361 IP Phone  MiVoice 5370 IP Phone  MiVoice 5380 IP Phone	• Intuitive and user-friendly menu prompting with Foxkey and central navigation key • All the system features can be used • Excellent voice quality due to Mitel Hi-Q™ wideband audio technology • Automatic update of the phone software • Connection via Ethernet • Powered via Ethernet (POE) or power supply • Wall mounting possible • Web configuration interface	MiVoice 5370 IP/MiVoice 5380 IP: • Expansion key modules can be connected • Headset socket with DHSG standard • Integrated switch for connecting a PC MiVoice 5380: • Backlit display • Optional Bluetooth module • Can be used as reception phone in hospitality environments • Can be used as operator console when combined with expansion key module
Note: The MiVoice 5360 IP IP system phone is supported as before.		

Tab. 6 Digital system phones of the MiVoice 5300 family

Product	Principal common features	Additional model-specific features
 MiVoice 5361 Digital Phone  MiVoice 5370 Digital Phone  MiVoice 5380 Digital Phone	• Intuitive and user-friendly menu prompting with Foxkey and central navigation key • All the system features can be used • Automatic update of the phone software • Connection via DSI interface • Two phones can be connected per DSI interface • Powered via DSI bus or power supply • Wall mounting possible	MiVoice 5370/MiVoice 5380: • Expansion key modules can be connected • Headset socket with DHSG standard MiVoice 5380: • Backlit display • Optional Bluetooth module • Can be used as operator console when combined with expansion key module

Tab. 7 Digital system phones of the Dialog 4200 family

Product	Principal common features	Additional model-specific features
 Dialog 4220  Dialog 4222  Dialog 4223	• Configurable number and function keys with LED • System features can be used via function codes • Hearing aid compatible • Connection via DSI interface • One phone can be connected per DSI interface • Powered via DSI bus or via optionally power supply • Wall mounting possible	Dialog 4222, Dialog 4223: • Graphics-compatible display • System features operated using menu prompting • Expansion key module(s) can be connected • Headset socket • Hands-free feature • Configurable team keys Dialog 4223: • 4 softkeys

Tab. 8 Cordless system phones of the Mitel 600 DECT family

Product	Principal common features	Additional model-specific features
 Mitel 612 DECT Phone  Mitel 622 DECT Phone  Mitel 632 DECT Phone  Mitel 650 DECT Phone	• Intuitive and user-friendly menu prompting with Foxkey and central navigation key • Colour display • All the system features can be used • Automatic update of the phone soft-ware • Backlit display and keyboard • Headset socket • Automatic handover and roaming • Can be operated on both the DSI radio units SB-4+, SB-8, SB-8ANT and the SIP-DECT® radio units RFP L32 IP, RFP L34 IP and RFP L42 WLAN	Mitel 622 DECT/Mitel 632 DECT/Mitel 650 DECT: • 3 configurable side keys • Vibra call • Bluetooth interface • USB Interface • micro-SD card interface • Power battery (optional) Mitel 632 DECT: • Complies with industry standard (IP65) • With emergency button and sensor alarms, suitable for personal pro-tection Mitel 650 DECT: • Supports the DECT standard CAT-iq (Cordless Advanced Technology – internet and quality) for high-quality broadband telephony (can be used with Mitel SIP-DECT only).
Note: The Mitel 610 DECT, Mitel 620 DECT, Mitel 630 DECT, Office 135/135pro and Office 160pro/Safeguard/ATEX cordless system phones are supported as before (not all system features can be used).		

Tab. 9 Analogue Mitel phones

Product	Principal common features	Additional model-specific features
 Mitel 6710 Analogue Phone  Mitel 6730 Analogue Phone	• Destination dialling keys • Frequency dialling or pulse dialling • Handsfree • Adjustable volume (handset and loudspeaker) • System features can be used via function codes • Headset connection • Wall mounting possible • Functions controllable via communi-cation server: Message display on/off, delete redial key memory. • Ideally suited for hospitality and hotel environments	Mitel 6730 Analogue: • Three-line display • 100 phone book contacts • 50 entries each on call list and redial list • Number/name display for incoming calls • Clock with wake-up function • Functions controllable via communi-cation server: Delete call lists and local phone book, set date, time and language.
Note: The Aastra 1910 and Aastra 1930 analogue phones are still supported.		

2. 5 Various phones, terminals and equipment

Thanks to the use of international standards other clients, terminals and phones, Mitel and third-party, can be connected and operated on the communication server:

- SIP-based phones
  With the integrated SIP protocol SIP-based phones (softphones, hardphones) - or via an SIP access point also WLAN and DECT phones - can be connected to the communication server. Besides the basic telephony functions, features such as call transfer, conference calls or CLIP/CLIR are also supported. Function codes can also be used to operate various system functions.
- Cordless phones
  The sturdy 9d DECT phones from the Ascom Wireless Solutions product portfolio can be logged on to the communication server as system phones. User-friendly messaging and alarm systems can thus be implemented in combination with the IMS (Integrated Message Server). Other DECT phones can also be operated in GAP mode.
- Analogue terminals
  All terminals (phones, fax, modem, etc.) approved by the network operator can be connected on the analogue terminal interfaces. The communication system supports pulse and frequency dialling modes.
- ISDN terminals
  ISDN terminals that comply with the Euro ISDN standard can be connected to the BRI-S terminal interfaces. The communication system provides a series of ISDN features at the S bus.
- Mobile/external phones
  Mobile/external phones can also be integrated into the communication system. They can then be reached under an internal call number, and their status is monitored and displayed. Internal/external calls can be made via the integrated mobile/external phone; system functions can also be executed using function codes. With the Mitel Mobile Client for mobile phones application all the main telephony functions are available with menu prompting (see "Mitel Applications", page 26).

2. 6 Solutions

- Alarming and Health care
  Thanks to the components Mitel Alarm Server, I/O-Gateway and the OpenCount application, flexible solutions are available for hospitals and old people's nursing homes. MiVoice Office 400 communication-server-integrated functions such as "Direct response" "Hotline alarm" or "PIN telephony" allow easy deployment of available features.

- Hospitality/Hotel
  The hospitality software package provides functions to implement a user-friendly accommodation and hotel solution in the range of 4 to 600 rooms. This solution is also ideally suited for the management of care homes and retirement homes. The functions are operated using the Mitel 6940 SIP, Mitel 6873 SIP, MiVoice 5380 / 5380 IP reception phone or the web-based Mitel 400 Hospitality Manager application. Reduced hospitality functionality are also available on Mitel 6920 SIP, Mitel 6930 SIP, Mitel 6867 SIP and Mitel 6869 SIP phones. Connection to a Property Management System (PMS) via the communication server's Ethernet interface is also possible. The commercially available FIAS protocol is provided for this purpose.
- Mobility
  Mobility solutions, especially Mitel Mobile Client (MMC), enable employees to log on to the company network using their mobile phones. The MMCC Compact and MMCC 130 controllers allow mobile users to move back and forth between the internal WLAN coverage and the mobile radio network without the call being interrupted. Moreover, with Mitel SIP-DECT and Mitel 600 DECT series phones comprehensive solutions can be provided for wireless telephony on IP-based networks. In so doing, RFP radio units are directly connected to the LAN like a VoIP device.

2. 7 Applications and application interfaces

A distinction is made among applications between Mitel-specific applications and certified applications supplied by third parties.

The Mitel applications Mitel Open Interfaces Platform (OIP) and Mitel 400 CCS run either on the integrated applications server or on a customer server. The fax service is offered on the integrated application server only. Certified third-party applications are always installed on a customer server. The applications on the customer server communicate with the communication server via standardised interfaces (see "Application interfaces", page 28).

Auxiliary applications for planning and the configuration and park management are available as a web application.

2. 7. 1 Mitel Applications

Tab. 10 Mitel applications

Application	Main features
Mitel Dialer	• Simple first party CTI application • Dial, answer, hang up • Integration in Outlook, Lync 2013 and Office 365 • Search in directories • Compatibility with MiVoice 5300, MiVoice 5300 IP, Mitel 6800/6900 SIP, Mitel 600 DECT series phones • Installation via SSP or WebAdmin • Click to call support (e.g. for Hospitality Manager)
Mitel Open Interfaces Platform (OIP)	• Application interface for deep integration of applications by Mitel or other manufacturers (see "Application interfaces", page 28) • Easy to manage through an integrated web-based application • Integrates the MiVoice 1560 PC Operator and Mitel OfficeSuite applications • Presence-controlled communication coupled with Outlook diary entries • Integration of contact databases and directories (Outlook, Exchange, Active Directory, LDAP directories, phone book CD) • Integration of building automation equipment and alarm systems • Call centre functions with flexible routing algorithms, skill-based agent groups and emergency routing • Unified messaging with notification whenever new voice messages are received via email (incl. message attachment) • Partner program for integrating and certifying applications by other manufacturers • Pre-installed on the applications card CPU2-S of the Mitel 470communication server. • Also available as OIP Virtual Appliance, for installation on a VMware server.
Mitel MiCollab	Comprehensive Unified Communications and Collaboration solution: • Central software provided for industry standard servers or virtual environments • Integration of Microsoft® Outlook®, IBM® Lotus Notes® Google®, Microsoft® Lync® etc. UC clients for desktop, web and mobile applications: • Comprehensive real-time presence information • Dynamic call distribution • Real collaboration with joint use of the desktop and documents • Easy retrieval of voice messages • Secure instant messaging (IM) and data transmission • Audio, web and video conferences
Mitel 400 CCS	• Mitel 400 CCS is an additional application for the Mitel 400 Call Center, and provides statistics / reporting functions and agent monitoring (CCS = call centre supervision). The licensing of the application is made via OIP. • Pre-installed on the applications card CPU2-S of the Mitel 470communication server.
Mitel OpenCount	• MitelOpenCount is a software package used for the call logging management on the communication system. It consists for selected sectors of basic, comfort and premium solutions and is installed on an external server.

Application	Main features
Mitel BusinessCTI	• Powerful Unified Communications solution • Presence management with calendar integration • Instant Messaging (chat), video, SMS and e-mail functions • Compatibility with the federation between Mitel Business CTI servers and/or Microsoft Lync and OCS • Easy integration into CRM and ERP systems • Compatible with other call managers • Clients for PC (Windows, Mac) and mobile phones/tablets (Android/IOS) available • Optional additional modules Mitel BusinessCTI Analytics
MiContact Center Business	• Contact Center on a location with up to 80 agents • Progress reports • Real-time monitoring • Dynamic agents and wait loop control • Screen pop • Intelligent Messaging • Multimedia compatibility
Mitel Border Gateway (MBG)	• Highly scalable solution which offers mobile and external workers secure and seamless access to the company's voice and data applications, regardless of their location. How to deploy such a solution refer to the document “Mitel SIP Teleworker via MBG on MiVoice Office 400”.
Mitel Alarm Server	• Specially designed for use in hospitals and nursing homes, industries and businesses as well as public domains. • Mitel Alarm Server monitors processes, activates the required services, sets off alarms based on predefined samples or notifies selected recipients via paging, e-mail, SMS or voice message. • The alarm can be set off via a nurse call or fire-alarm system (ESPA interface), via a key predefined on the Mitel DECT or system phone, an alert button, web client, or by calling the alarm server (audio guide), or via e-mail (subject line analysis).
Fax service	• The server-based fax service integrated on the CPU2-Sapplications card converts incoming messages into PDF files and sends them to the recipient as an e-mail attachment. When outgoing PDF files in e-mail attachments are converted into fax messages. Fax messages can also directly be sent from MS applications via a special printer driver. • Pre-installed on the CPU2-S applications card of the Mitel 470communication server.

Tab. 11 Planning and configuration applications

Application	Main features
Mitel CPQ	• Web-based planning application for Mitel communication platforms (CPQ = Configuring Planning Quoting) • Uses project data to calculate the necessary communication server complete with terminals, interface cards, modules and licences • Country-specific adaptations possible for accessories • Stored price lists and configurable quote compilation • No installation necessary

Application	Main features
WebAdmin	• Web-based configuration tool for configuring and monitoring a single system or an entire network (AIN) • Access control with user accounts and predefined authorization profiles • Special accesses for hospitality solutions • Integrated online help and configuration assistant • Integrated in the communication server software package
Mitel 400 Hospitality Manager	• Integrated web-based application used to operate functions in the hospitality sector • List view and floor-by-floor view of the rooms • Functions such as check-in, check-out, group check-in, notification, wake-up call, retrieval of call charges, maintenance list, etc.
Self Service Portal (SSP)	Web-based application for end-users, which allows personalised configuration of a telephone: • Functions key assignment and printing of labels • Setting the idle text and language • Setting the presence profiles, personal call routing, voice mail, forwarding, etc. • Setting up dial-in conference rooms • Creating private phone book contacts • Managing personal data such as e-mail address, password, PIN, etc.
Secure IP Remote Management (SRM)	• Server-based solution for secure IP remote management • No router and firewall configuration or VPN connection setup required • Allows configuration via WebAdmin once the connection has been set up • No installation necessary

2. 7. 2 Application interfaces

The most important interface for own and third-party applications is the interface of the Mitel Open Interfaces Platform (OIP). This open interface allows the applications to be deeply integrated with telephony. Third-party applications can also be integrated on MiVoice Office 400 series systems via different interfaces without OIP.

2. 7. 2. 1 Mitel Open Interfaces Platform

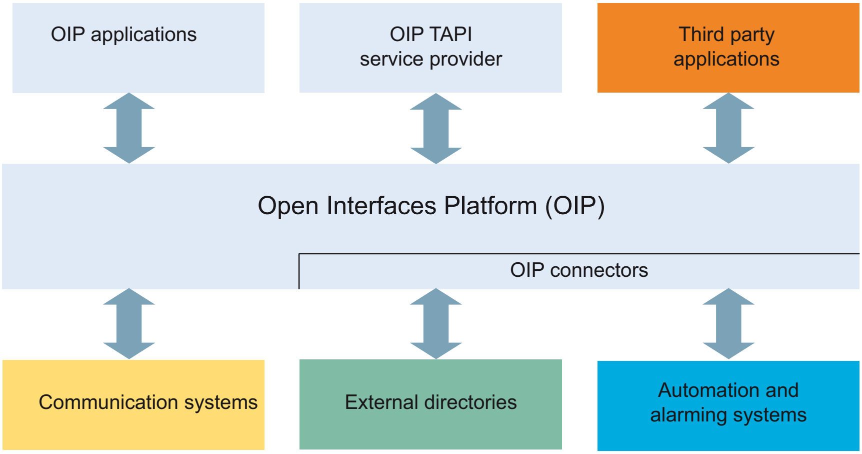


Fig. 3 OIP as middleware between communications system, external data sources and applications

OIP services

The OIP services are the central components of OIP. They are used to control the system and make the OIP features and interfaces available. Thanks to the modular organisation and vast configuration possibilities, versatile and customer-specific solutions can be set up.

OIP applications

Sophisticated Softphones are available as OIP applications and are controlled as clients via OIP.

- Mitel OfficeSuite is a rich-client application, which significantly broadens the range of functions of the coupled fixed and cordless phones.
- MiVoice 1560 PC Operator is an operator application which can be used as rich-client application together with a fixed or cordless phone or alone as Softphone.

Possible OIP application fields are listed in the following sections:

OIP as directory server

Already available directories, databases and phone books are linked to OIP and made useful for name dialling and identification.

Integration is compatible with many standard databases such as Microsoft Exchange, Microsoft Outlook, Microsoft Active Directory, communication server directories, LDAP and ODBC directories and electronic phone books.

Moreover, Microsoft Exchange directories can be directly synchronised.

Unified Communications - OIP as telephony server

When OIP is used as telephony server, telephony integrates in a scalable manner into IT communication: Top-class Softphones, PC-operated fixed and cordless phones, presence-controlled call, voice mail control and calendar coupling via presence profiles, name dialling and call number identification via all linked company directories, synchronisation of Microsoft Exchange contacts, e-mail notifications, etc. facilitate daily communication.

OIP as operator centre

Several multi-functional operator applications can be organised with call centre functions in operator groups.

OIP as Free Seating Server

OIP supports and expands the MiVoice Office 400 free seating function: A user logs on at a free seating workstation and the phone automatically takes over his call number and device configuration.

OIP as call center

The powerful Mitel 400 Call Center is an integral part of OIP and provides all the main features such as flexible routing algorithms (cyclical, linear, longest time available, CLIP-based, last agent), skill-based agent groups as well as an analysis of the call centre data (online and offline) with chart-based evaluation. In the event of a network interruption the emergency routing ensures the maximum availability of the system.

The agent functionality is available on all system phones including Softphones. This applies equally to home workstations and to all the users on a Mitel Advanced Intelligent Network. The one number user concept can also be set up for agents, which provides the staff of a Call Center with maximum mobility within the company.

The Mitel 400 Call Center is easy to manage and configure thanks to OIP WebAdmin. Various monitoring functions, simple statistical evaluations and work group control can be comfortably implemented using the administration interface.

Mitel 400 CCS is an extension of the Mitel 400 Call Center and offers several possibilities of statistically evaluating the call centre operation. Offline and online reports enable the call center operator to analyse and optimise call centre operations.

OIP as application interface

Certified third-party manufacturers can, for instance, integrate sector-specific applications into the MiVoice Office 400 and OpenCom communication environment.

OIP as automaton and alarm system

External alarm systems and building automation equipment (e.g. KNX) are easily monitored through the connection to the communication system. This allows information to be exchanged in a simple way between the systems. In this way the user can use his system phone for voice communications and for monitoring external systems.

The I/O service offers a wide range of features which allows very flexible uses and versatile applications. Some of its examples are listed below:

- Alarming equipment for maintenance personnel
- Monitoring of production processes
- Forwarding messages as e-mails
- Connection to building automation systems (KNX)

With the graphical interface (tree structure) events and the relevant actions are easily linked with one another.

OIP in a networked environment

An OIP server can also be used in an AIN. To do so, it will be linked to the Master. In addition, several communication systems can also be connected to an OIP server. It is then possible for instance to obtain network-wide call logging for all the systems, to display call charge information on the system phones or to display status in the presence indicator field of a PC operator console for all the users connected.

See also:
More information can be found in the Mitel Open Interfaces Platform system manual and in the OIP WebAdminOnline help.

2. 7. 2. 2 Message and alarm systems

MiVoice Office 400 supports several message formats and message protocols for implementing messaging, monitoring and alarm systems.

Internal messaging system for system phones

The internal messaging system for system terminals allows users to exchange predefined or user-defined text messages between system phones. Text messages can also be sent to individual users or message groups.

The internal messaging system does not have an interface with which it can be addressed directly. However it can also be operated via OIP.

External messaging, monitoring and alarm systems

The powerful ATAS/ATASpro protocol is available via the communication server's Ethernet interface for applications in the security and alarming sector. This protocol can be used to implement customised alarm applications. An alarm appears on the display of system phones, complete with the freely definable user functions that apply only to that alarm. In addition the duration of the tone as well as its volume and melody can be freely defined by the user for each alarm.

The Mitel Alarm Server is a flexible solution which can be used in all sectors to process and record alarms. It can be used, for instance, in old people's nursing homes and assisted-living homes, as well as in other different facilities such as hotels, industrial plants, shopping centres, schools or administrations. When used together with Mitel SIP-DECT it is even possible to dynamically determine the environment of the alarm solution using the location feature provided by the DECT system.

The cordless DECT phone Mitel 630 DECT is specially designed for applications in the security and alarming sector. Besides a special alarm button it also features a man-down alarm, a no-movement alarm and an escape alarm. Sensors inside the phone constantly check the handset’s position and motion. An alarm is triggered if the phone remains in a virtually horizontal position or motionless for some time or if the handset is shaken violently.

2. 7. 2. 3 CTI - Computer Telephony Integration

The Computer Telephony Integration (CTI) integrates telephony services in the company process. Besides conventional telephony features Mitel Open Interfaces Platform (OIP) offers many other convenient functions, which supports the employees with their daily work, for instance:

- Dialling by name for outgoing calls and CLIP display for incoming calls offers an added value by the integration of external directories and databases.
- Notification of Microsoft Outlook appointments on the system phones
- Presence-controlled communications with Busy Indicator
- Automatic Call Distribution
- Access to system configuration, what a maximum integration of different systems ensures

And of course the communication system supports also First and Third-Party CTI interfaces for commercial CTI applications based on the Microsoft TAPI 2.1 standard.

Terminal supervision/control on the communication server by third-party applications via the CSTA protocol is also supported.

First-party CTI

A first-party CTI is the direct physical connection between a phone terminal and a telephony Client (workstation PC). Telephony functions and telephone states are controlled and monitored on the telephony Client. A first-party CTI solution is ideal for a small number of CTI workstations and is easily implemented.

MiVoice Office 400 supports First-Party CTI on all system phones via the Ethernet interface. For some applications (e.g. Office eDial) the First-Party TAPI Service Provider (AIF-TSP) is required. Other applications (e.g. Mitel Dialer) use the CSTA protocol.

Application example

- Dialling from a database (phone book CD, etc.)
- Caller identification (CLIP)
- Creating a call journal
- Mitel Dialer (see Tab. 10, page 26)

Third-party CTI

Third-party CTI is an user-friendly multi-station solution. In contrast to first-party CTI, third-party CTI controls and monitors several system phones (including cordless phones) via the central telephony server, which is connected with the communication server. In addition phones on ISDN and analogue interfaces can also be monitored. PC and phone allocation is handled by the telephony server.

The third-party CTI connection is effected via Ethernet using the Mitel Open Interfaces Platform (OIP). To this end the OIP is installed on the telephony server. Third-party connections via Ethernet with CSTA are also possible.

Application example

- Busy indicator
- Group functionality
- Networked CTI solution
- Automatic Call Distribution (ACD)

2. 7. 2. 4 ISDN interface

MiVoice Office 400 supports the ISDN protocols ETSI, DSS1 and QSIG.[1] Besides the possibility of networking various systems into a PISN (Private Integrated Services Network) via the ISDN interface, these protocols also provide various functions that can be used for connecting external applications (e.g. IVR systems, fax server, voice mail systems, unified messaging systems, DECT radio systems).

1) for USA and Canada on Mitel 470 other protocols are supported.

2. 7. 2. 5 Configuration

The MiVoice Office 400 communication server is configured via the web-based WebAdmin application. Other components of the application include special accesses for hospitality and hotel solutions as well as a configuration wizard.

2. 7. 2. 6 System monitoring

The system status is monitored with event messages which can be sent to various internal or external destinations. Examples of message destinations are: system phones, events log (WebAdmin), e-mail recipients, SRM servers, alarm servers (ATAS) or SNMP destination. Event messages are also accessible via the Mitel Open Interfaces Platform for application manufacturers.

2. 7. 2. 7 Call logging

The Call Logging Manager includes data acquisition for incoming traffic (ICL), outgoing traffic (OCL) and the counting of the acquired call charges according to a variety of criteria. The data can be retrieved via different interfaces and subsequently processed.

2. 7. 2. 8 Hospitality/Hotel

The MiVoice Office 400 communication servers offer you several possibilities to implement a hospitality and hotel solution, with different operation applications and interfaces. Configuration is done through WebAdmin. The Mitel 6940 SIP, Mitel 6873 SIP, MiVoice 5380 / 5380 IP reception phone or the web-based Mitel 400 Hospitality Manager application is available to operate the functions. Reduced hospitality functionality are also available on Mitel 6920 SIP, Mitel 6930 SIP, Mitel 6867 SIP and Mitel 6869 SIP phones. A connection to a Property Management System (PMS) via the communication server's Ethernet interface is also possible. The commercially available FIAS protocol is provided for this purpose.

2. 7. 2. 9 Voice over IP

MiVoice Office 400 is a native VoIP solution. Apart from the possibility to operate IP system phones and SIP phones via the Ethernet interface, MiVoice Office 400 systems can also be networked over IP.

2. 8 Connection options

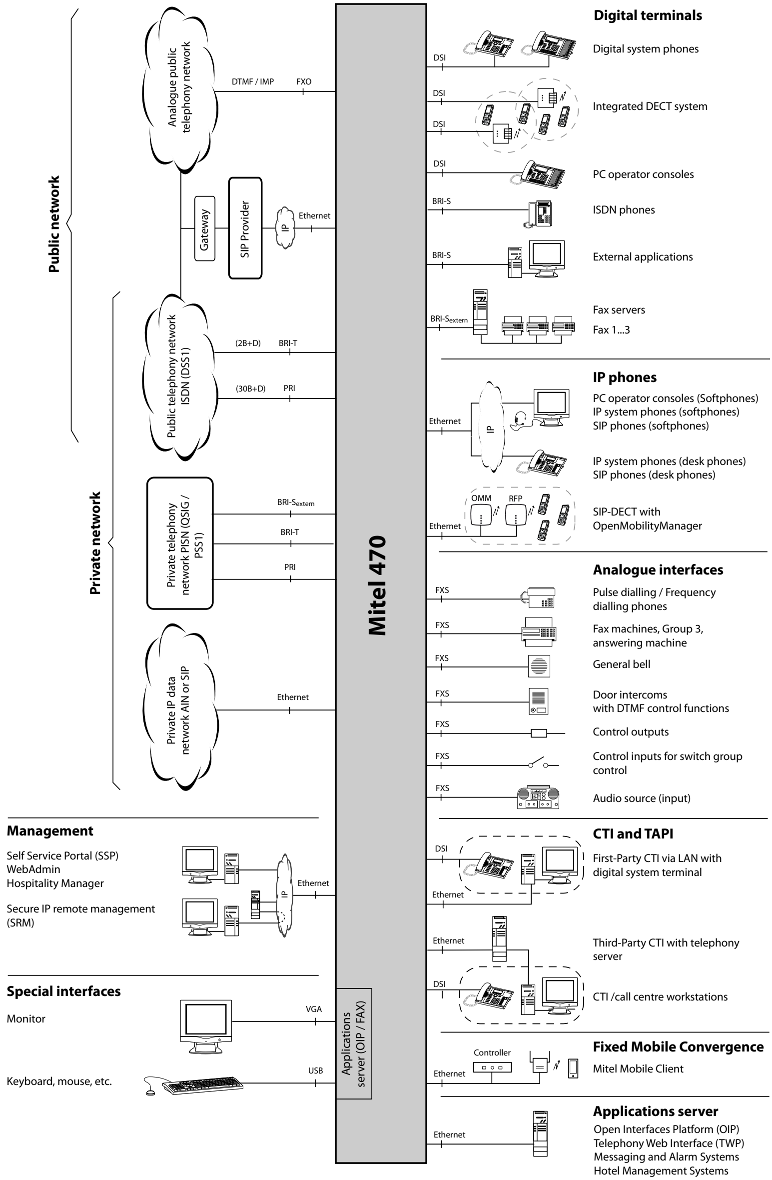


Fig. 4 Overview of interfaces with possible terminal equipment

2. 9 Getting started

If you are setting up an MiVoice Office 400 communication system for the first time, it may be useful to set up a test system step by step on site.

After working through the following chapters you can make internal calls between the different types of phones connected to the server. Furthermore you will have a perfect configuration platform to learn more about the system, its features and expansion possibilities.

2. 9. 1 General requirements

You need a Windows OS computer with internet access, the *System Search* application and access to Mitel Connect.

If you plan to address the communication server with a static IP address (recommended) you may obtain it from your IT administrator.

In order to allocate your IP and SIP phones to the communication server, DHCP service should be available in your subnet. (Your communication server has an integrated DHCP server as well, however it is switched off as per default.)

If you plan to set up a SIP trunk, you need a SIP account by a SIP provider of your choice.

Required accesses

The URL's listed below refer to proprietary Mitel sites. You need a partner login to access them. If you do not have a Mitel partner login, ask your sales partner for more information.

Tab. 12 Mitel sites you need access to:

	Title	
[1]	MiVoice Office 400 DocFinder or Mitel eDocs	www.mitel.com/DocFinder or Mitel eDocs
[2]	Access to Mitel Connect (for *Mitel CPQ*, *Licences & Services* and *Software Download Center*)	https://connect.mitel.com

Required tools

- Torx screwdriver T10 and T20
- Phillips screwdriver size #1

2. 9. 2 Plan and order

Set up your MiVoice Office 400 project in Mitel CPQ first. As a result, you will obtain a list of needed components, a slot usage layout, a DSP configuration table and a licence overview.

Mitel CPQ is designed to support you with the different activities in the sales and ordering process. It is a web-based application for online usage. You can access the application through the Mitel Connect Portal [2].

Save the component list either as Microsoft Excel or Word file and place an order with your Mitel reseller.

2. 9. 3 Download documents, system software and tools

Before you start, download the documents and applications from the proprietary Mitel sites.

Proceed as follows to organize all downloads in a common folder:

1. Download the *Documentation set* from the Mitel document portal [1], double-click the file and follow the installation wizard steps.
2. Choose *My Documents* or another suitable target directory and install the *Documentation set*. A folder named *Mitel* is created automatically.
3. Download the latest system software from [2] into the same folder and double click the file. The software (zip) and the release notes (pdf) will also be extracted to the folder named *Mitel*.
4. Download the latest System Search application from [2] into the folder named *Mitel*. The application needs no installation and can be executed by a double click.

2. 9. 4 Equip, connect and power on

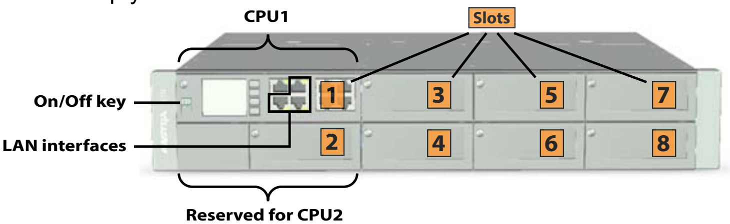
The communication server ships with a plugged-in processor card (CPU1) containing some interfaces and is ready to use as a basic system.

CAUTION!

- Before you begin, read the product information and safety instructions carefully (see the PDF included in the *Documentation set* or the printed sheet included in the delivery package).
- To prevent ESD damage to the components, always touch the earthed metal case of the communication server before carrying out work inside the housing. This also applies to handling interface cards, processor cards, and system modules that are not packed inside the ESD protective wrapping.

1. Ensure that the communication server is disconnected from the power supply.

2. Insert interface cards (if any) starting with slot 3 and tighten the screw on them on. Leave slot 2 empty.



3. Install system modules (DSP modules or an EIP module) if any:
   - Remove the CPU1 card.
   - Mount the system modules on the CPU1 card.
   - Insert the CPU1 card back in slot 1 and tighten the screw.
4. Connect the LAN cable to one of the LAN interfaces on the front panel.
5. Set the voltage converter on the rear panel to the voltage of the available mains power (230 VAC or 115 VAC).

CAUTION!
Printed circuit boards may be damaged or become defective if the communication server is operated at a voltage different from that set on the voltage selector.

6. Connect the power plug to the socket on the rear panel and to the power supply.

CAUTION!
Make sure all the housing openings of the communication server are closed during operation to ensure a controlled flow of air.

7. Start the communication server by pressing the On/Off key on CPU1.

   When the start-up is complete, the communication server runs in normal operating mode. The status LED above the On/Off key flashes green. DHCP is switched on by default.

2. 9. 5 Put into operation

Search for the communication server in the IP network

1. Connect your computer to the IP network ensuring that your computer is in the same subnet as the communication server.
2. Double-click *System Search* to start the application.
3. In *System Search*, click *Search*.

   All communication servers in the same subnet are listed.

Tip
If your communication server is not listed, your computer is in another subnet. If it is not possible to connect the IP network in the same subnet, connect your computer to the communication server either directly or through a switch. Click *Search* again.

Set the IP address data of the communication server

1. In *System Search*, select your communication server from the list and click the *IP settings* tab.
2. Set *DHCP* to *Off*, enter the static IP address of the communication server and the corresponding *Subnet mask*. Click *Save*.

   The password window opens.

   Note
   Although you can use DHCP, we recommend that you address the communication server using a static IP address.
3. Enter the default user name and password (*admin* / *password*) of the communication server and click *OK*.

   A message appears that the IP address has been successfully changed.

Initialize and localize the communication server

1. In *System Search*, click *Search*.

   Your communication server is now listed with the new IP address.
2. Select your communication server from the list and click *Configure*.

   WebAdmin opens in your web browser and shows the *Sales channel selector* view.
3. Select your *Sales channel*.

   Note
   You must select the correct *Sales channel* as it is mapped to the licence code.
4. Click *Next*.

   The *Software update* view is displayed. We recommend that you update the communication server to the latest software release.
5. Choose the *Manual software upload* entry and upload the system software that you have already saved to your hard disc (see chapter "Download documents, system software and tools", page 37”). During the software update (or if you chose not to update the software, after you click *Next* in step 6) a first start is executed to set the sales channel and the country specific settings.
6. Click *Next*.

   The *Upload audio guides* view is displayed. The communications server uses spoken text for several purposes like voice mail, presence information or auto attendant. These texts are stored in audio files. You can download audio guide languages through the menu *Localize* in *System Search* and then upload them to the communication server in this view.

Note
If your communication server has Internet access, you can choose to skip this step, because you can download the audio guide languages later from a Mitel FTP server through the *Localization* (*=e6*) view in WebAdmin.

7. Click *Next*.

   The *First access* view displays, prompting you to change the default password of the administrator account, to choose the *System language*, and to enter a *Site name*.
8. Click *Next*.

   The first page of the WebAdmin *Setup wizard* opens.

Configure the basic settings using the Setup wizard

Tip
If you need help while going through the steps of the wizard, click *Help* in the upper right of the *Setup wizard*.
A new help window appears. You can leave the help window open, while going through the steps.

1. On the first page of the *Setup wizard*, you register or activate the communication server by uploading a valid *Licence file*.
   - Copy the *Equipment ID (EID)* to the clipboard.
   - In a new browser window, log in to the Mitel Connect portal [2] and open the *Licences & Services* section.
   - Option 1: If you have a voucher, enter the voucher number in the *Voucher edit field*, click *Register Voucher* and follow the instructions. You need to enter the *Equipment ID (EID)* during the procedure. On completing the procedure, you will obtain a *Licence file*.
   - Option 2: If you have no voucher, enter the *Equipment ID (EID)* in the *Activate product* edit field, click *Activate product* and follow the instructions. On completing the procedure, you will obtain a *Licence file*.
   - Upload the *Licence file* in the WebAdmin *Setup wizard*.
     Your communication system is now registered and activated.
     The new licences are enabled. You can see them on the licence overview page.

Note
If you do not activate the communication server, it will switch to a restricted operating mode after four hours.

2. Click *Apply and Next*.

   The second page, *Setting up the IP addressing*, opens.
   Set the *Gateway* address and a *Primary DNS server*.

   Note
   If you do not set these parameters, you cannot load audio guides or update Mitel SIP phone strings from the Mitel download server.

3. Click *Apply and Next*.

   The third page, *Configuring media resources*, opens.
   On this page, the system proposes to configure the DSP resources automatically. You can use this configuration to begin with. You can always change the DSP settings under *Configuration* - *System* - *Media resources* (*=ym*). Check the options for FoIP and DECT resources, if applicable.

4. Click *Apply and Next*.

   The fourth page, *Setting up the numbering plan*, opens.
   This page displays the predefined call numbers of the internal numbering plan. You can edit or delete these numbers.

5. Click *Apply and Next*.

   The fifth page, *Setting up SIP providers*, opens.
   This page allows you to set up a SIP provider profile or import a predefined SIP provider profile from an XML file. If your communication system will not be connected to the public network through a SIP provider, skip this step.

6. Click *Apply and Next*.

   The sixth page, *Setting up users, terminals and DDIs(DIDs)*, opens.
   On this page, you set up users, terminals and DDIs(DIDs).

7. Click *Apply and Next*.

   The seventh page, *Setting up the auto attendant*, opens.
   This page allows you to set up an auto attendant, if needed. The auto attendant enables you to specify, what options are offered to a caller while greeting the caller. The caller can select any of the options by dialing a single digit.

8. Click *Apply and Next*.

   This completes the setup. Click *Restart* for the configurations to take effect.

2. 9. 6 Register and connect the phones

As you allocated phones to users in step 6 of the Setup wizard, the data instances for the phones have been automatically created. In this part of the procedure, for registering the phones, you pair the data instances with the physical phones.

Note
Mitel SIP phones get their time and date from an NTP server. To ensure this, enable the *NTP service* in *System* / *General* (*=ty*) and enter the IP address of the NTP server.

Register a Mitel SIP phone

1. Go to *Terminals* / *Standard terminals* (*=qd*) in WebAdmin and click the phone you want to register with the communication server.
   The automatically generated SIP credentials and registration credentials (*Registration user name* and *Registration password*) of the phone are displayed. You will need to provide the registration credentials later to register the phone.
2. Add one or more expansion key modules to the phone, if available.
3. Connect the phone to the IP network and to the power supply by using the optional power adapter. If your IP network supports PoE, no power adapter is required.
4. Restart the phone.
   The phone searches for the communication server. If more than one communication server is available, the phone lists them in the format <XXX–MAC address>.

   Tip
   You will find the MAC address of your communication server in *IP network* / *IP addressing* (*=9g*) of WebAdmin.
5. Choose your communication server from the list, and when prompted, enter the *Registration user name* and the *Registration password*.

   The phone registers with the communication server. If a new phone software is available, the phone automatically updates and restarts.

Register a MiVoice 5300 IP system phone

1. Add one ore more expansion key module(s) to the phone.
2. Connect the phone to the IP network and to the power supply using the optional power adapter. If your IP network supports PoE, no power adapter is required.
3. On the phone, keep the C-key pressed down to access the local *Administration* menu.
4. Set the static IP address of the communication server (*Administration* / *PBX settings* / *PBX address*). To change the settings you have to enter the administrator password first (default = *0000*).
5. Restart the phone and enter the call number of the user you want to allocate to this phone as *Registration code*.
   → The phone registers on the communication server. If a new phone software is available, it is automatically updated and the phone restarts again.

Connect the digital system phones MiVoice 5300

1. Add one ore more expansion key module(s) to the phones.
2. Connect the phones to the DSI interfaces on the front panel. Connect the phones in the same order as you have set them up in the previous chapter and start with the lowest port number.
3. The phones are registered and allocated to their phone data instance in the communication server. If you keep the suggested order, the phone type matches with the configured terminal type. You can fix a terminal mismatch in the WebAdmin *terminal* view.

Test your configuration

Now you are able to make internal calls between the phones you connected to your communication server. Do some calling tests between the different phone types and check the audio. In the documentation set you will find the user's guides to your phones.

2. 9. 7 Make further configurations

Congratulations, you have set up the communication server for self training purposes. Now you have a perfect configuration platform to learn more about the communication server, its features and expansion possibilities.

For further configurations, use the *WebAdmin configuration assistant* and the online help. For detailed information, see the user's guides and system manuals (part of the *Documentation set*).

3 Expansion Stages and System Capacity

The basic systems can be expanded using interface cards, system modules, an applications card and licences. The expansion possibilities available and the maximum system capacities need to be known so the communications system can be ideally adapted to customer requirements. With the project data the optimum hardware configuration is easily determined using the project planning application Mitel CPQ.

3.1 Summary

Expansion possibilities for the Mitel 470 basic systems at a glance. The interface cards are fitted from the front into one of a total of 7 slots. System modules are fitted either to the call manager card or to interface cards. System modules are also used on other platforms: The DSP modules with Mitel 415/430 and the IP media modules with MiVoice 5000.

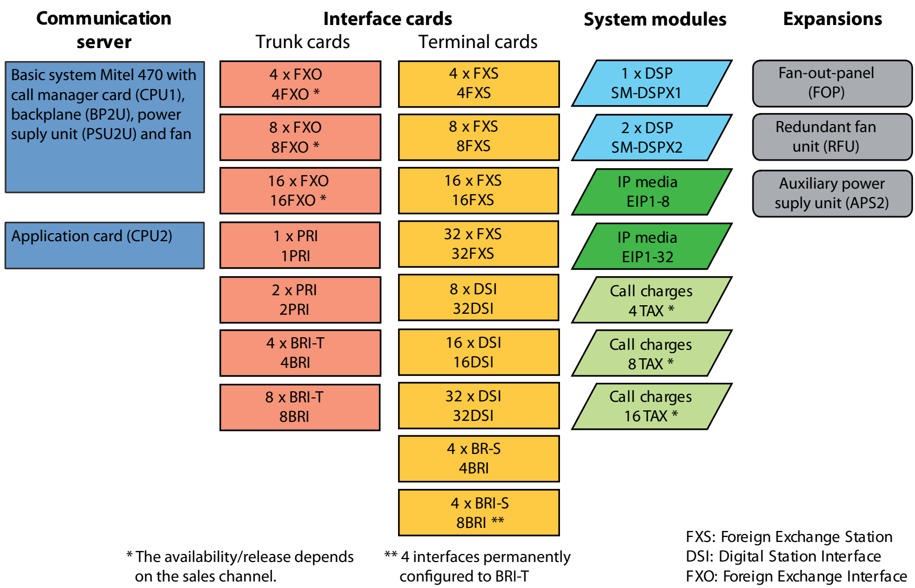


Fig. 5 Overview of the expansion possibilities

The basic system Mitel 470 can be expanded not just with interface cards and system modules but also with an applications card (CPU2). The applications card is supplied with pre-installed operating system, unified communications and multi-media applications.

The front-side RJ45 sockets of interface cards with 16 or more interfaces are partly or all four-fold assigned. With the FOP fan-out-panel they can be split again to individual sockets.

The Mitel 470 basic system has an integrated fan. The operating reliability of the communication server can be increased by fitting an optional redundant fan unit.

It is powered by an internal power supply unit (PSU2U). An external auxiliary power supply unit (APS2) is required for expansions involving a large number of power-consuming terminals. The auxiliary power supply unit also serves to increase the operating reliability. If the internal power supply unit fails, the external auxiliary power supply unit takes over the power supply.

3. 2 Basic system

The Mitel 470 basic system consists of the following components:

- Metal housing (2 height units) suitable for installation in a 19" rack or for desktop installation.
- CPU1 call manager card, fitted with a Flash card, a RAM module and an EIM card.
- 7 expansion slots with dummy covers fitted
- BP2U backplane fitted to electrically connect processor cards and interface cards.
- Fitted PSU2U power supply unit
- Fitted fan
- Power cord
- Rack assembly material

Fig. 6 Mitel 470 basic system

For electrical and thermal reasons the dummy covers must always be fitted. They are removed only to expand the basic system with interface cards or an application card.

For a clearer overview the figure below shows the open communications server from above with an additional fan fitted. The housing cover is in two parts. The upper, rear cover must be removed for the purpose of fitting an additional fan (see "Fitting an additional fan", page 100 for the procedure).

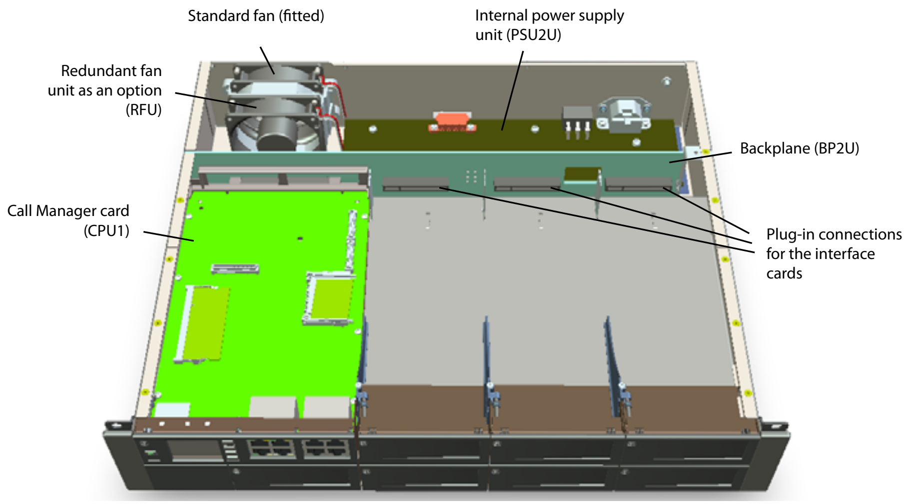


Fig. 7 Mitel 470 basic system fitted with a redundant fan unit

3. 2. 1 Interfaces, display and control elements

The interfaces accessible from the outside are located on the front and rear side of the basic system. The housing cover only needs to be opened when fitting an additional fan (see "Fitting an additional fan", page 100).

Basic system (without call manager card)

The figure below shows the positions of basic system interfaces without call manager card.

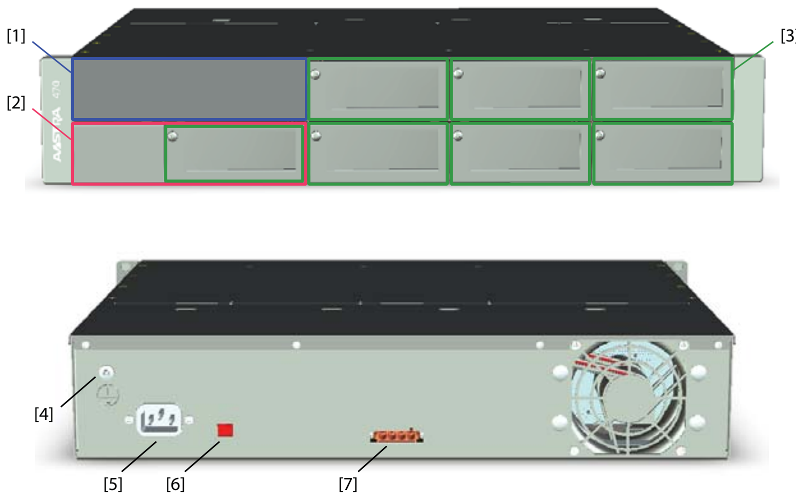


Fig. 8 Position of the interfaces on the basic system

Tab. 13 Interfaces of the basic system

Interfaces	Number of entries	Position	Remarks
Slot for call manager card CPU1	1	[1]	Device ships already equipped
Slot for application card CPU2	1	[2]	Can be fitted as an option
Slots for interface cards	7 1)	[3]	Can be fitted as an option
Interface for redundant fan unit	1		Connectors inside the housing
Earth connection	1	[4]	
Mains socket for 115/230 V power supply input	1	[5]	
115/230 V voltage converter	1	[6]	
Socket for auxiliary power supply unit APS2	1	[7]	

1) 1 fewer slot if CPU2 application card is fitted

Call manager card CPU1

The call manager card is the core the basic system and already fitted on delivery. Besides a powerful processor it also comprises a RAM module, a Flash memory card with the call manager software and an EIM card, on which some system related data are stored.

The call manager card comprises two powerful DSP chips, one of which can be assigned selectable functions. Two DSP modules can also be fitted as an option to further boost the media resources (see also "Media resources", page 51).

An IP media module can be fitted as an option to increase the number of VoIP channels (see also "IP media module", page 60).

Three individually configurable Gbit Ethernet interfaces are available on the front panel of the call manager card. The status of the interfaces is visible directly on the interfaces themselves thanks to the LEDs (see also "Ethernet interfaces", page 160).

Analogue voice and data terminals are connected via FXS interfaces. The call manager card comprises four of these configurable multifunctional interfaces (see also "FXS terminal interfaces", page 148).

The most striking display element on the call manager card is the backlit 1.8” colour display with the four navigation keys as control elements. It is used to display event messages or to execute maintenance functions. If the colour display is not available (e.g. during call manager system setup) the call manager status is indicated using the multi-coloured status LED on the On/Off button (see also "Call-Manager display and control panel", page 218).

The figure below shows the positions of the interfaces and of the display and control elements on the call manager card.

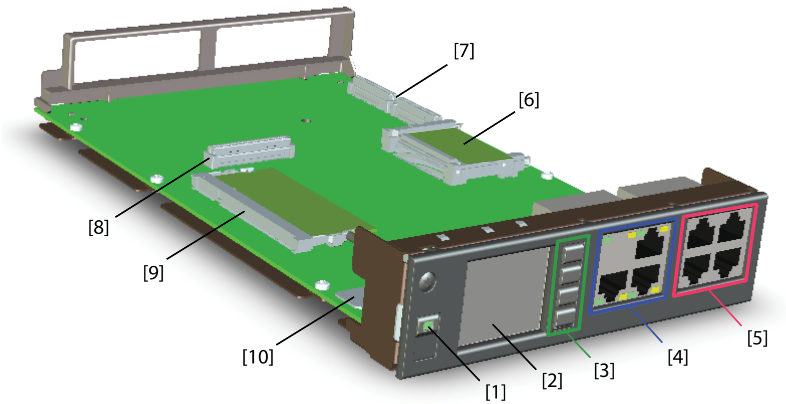


Fig. 9 Interfaces, display and control elements of the call manager card CPU1

Tab. 14 Interfaces, display and control elements of the call manager card CPU1

Interfaces, display and control elements	Number of entries	Position	Remarks
On/Off button with integrated status LED	1	[1]	
Colour display	1	[2]	
Navigation keys	4	[3]	
Ethernet interfaces 1Gbit/s (LAN)	3	[4]	RJ45 sockets
FXS terminal interfaces1)	4	[5]	RJ45 sockets

Interfaces, display and control elements	Num-ber of entries	Position	Remarks
Slot for Flash card	1	[6]	Device ships already equipped
Slot for DSP modules	2	[7]	Can be fitted as an option, stackable
Slot for IP Media module	1	[8]	Can be fitted as an option
Slot for RAM module	1	[9]	Device ships already equipped
Slot for EIM card	1	[10]	Device ships already equipped

1) Multifunctional analogue interfaces

3. 2. 2 Power supply

Internal power supply unit PSU2U

The Mitel 470 communication server is powered as standard directly with a mains cable. The voltage converter needs to be set to the correct position to match the mains power (230 VAC or 115VAC) (see also "Powering the communication server", page 106). The internal power supply unit PSU2U powers all the system components and a limited number of connected terminals.

External auxiliary power supply APS2

The external auxiliary power supply APS2 is used for the following purposes:

- Increasing the supply power available. This is required only for systems which are to operate a large number of terminals without their own power supply.
- As a redundancy for the internal power supply unit PSU2U. If either the internal or the external power supply unit fails, the system switches over to the intact power supply, without interruption.

The external auxiliary power supply APS2 is also powered by the 115/230 V mains.

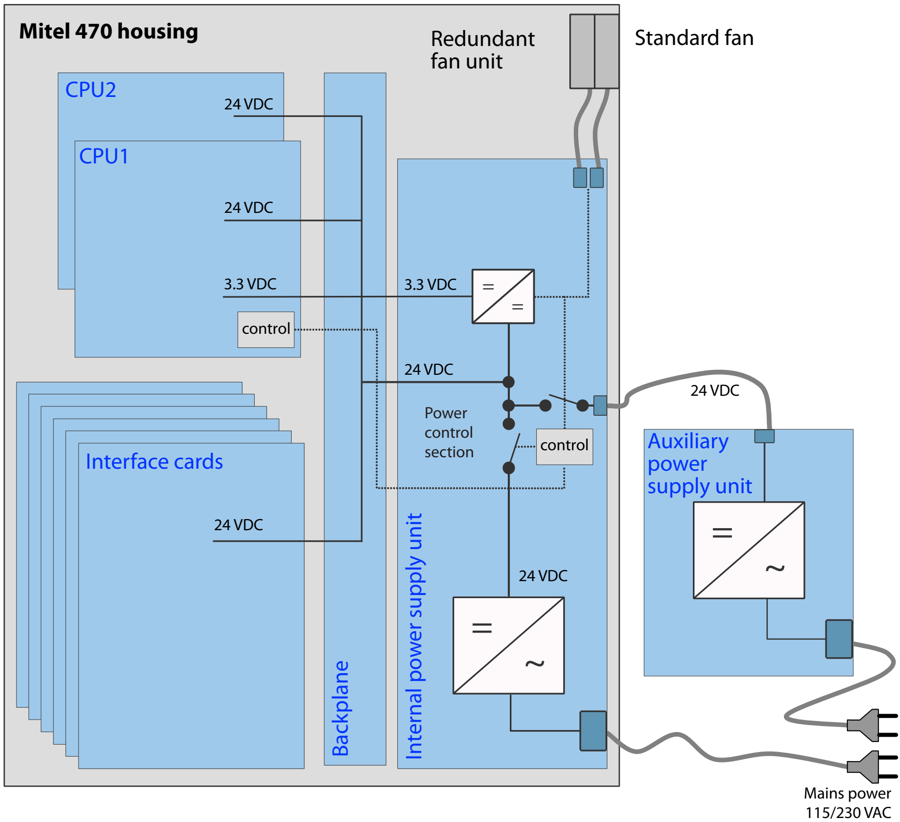


Fig. 10 Overview of the Mitel 470 power supply concept

Notes

- It is also possible to operate the communication server with the external power supply unit APS2 only In this case redundancy operation is of course no longer possible.
- To ensure that its operation is maintained even in the event of a mains outage, an external uninterruptible power supply (UPS) must be used.

See also:

For the available power outputs using the various types of power supply and for connecting the power supplies, see "Powering the communication server", page 106.

3. 2. 3 Ethernet concept

Mitel 470 provides three GBit Ethernet interfaces, which are routed to the front panel of the call manager card. They are used to connect to the customer's data network (LAN) and e.g. the IP connection with an SIP provider. The socket marked "WAN" currently has no function and remains covered.

Likewise the Ethernet interface on the front panel of the applications card is not used as the applications server is accessed via the WebAdmin configuration tool.

As the following schematic diagram shows, all the cards are internally connected with one another via Ethernet.

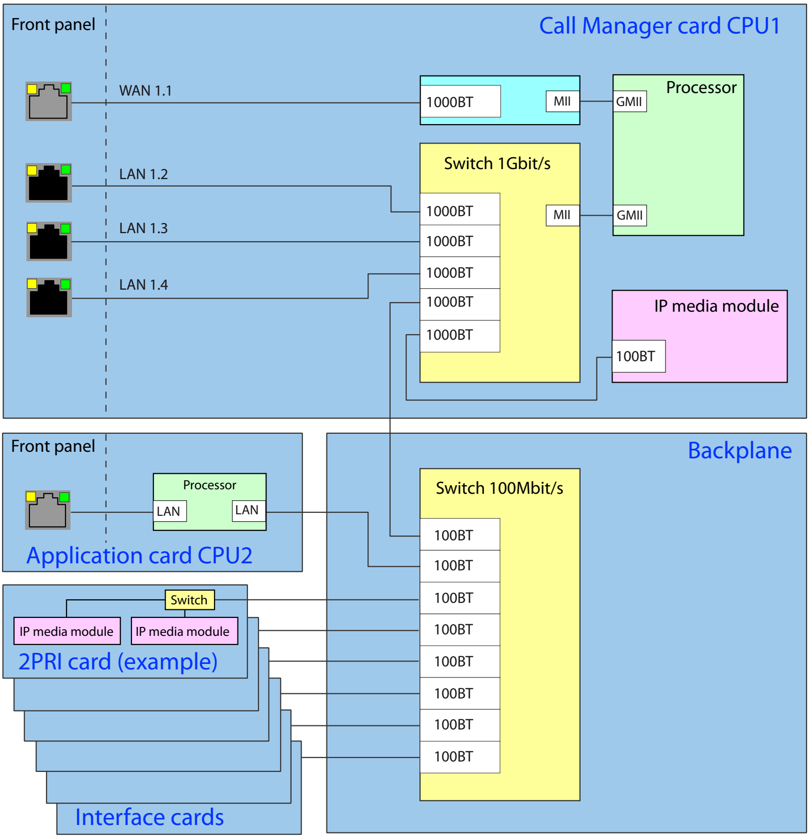


Fig. 11 Overview of the Mitel 470 Ethernet concept

3. 2. 4 Media resources

Media resources are used for complex signal processing functions, and made available by DSP chips. (DSP stands for Digital Signal Processor). They provide functions for conference circuits, DTMF sender and receiver, compression of voice data, etc. Two DSP chips are permanently fitted to the call manager card.

A DSP chip on the call manager card is allocated to fixed functions, which can be used without licences (see Tab. 15).

The functions of the second DSP chip can be selected to suit requirements. The functions are partly subject to licence (see Tab. 20).

The basic resources of the communications server can be expanded by fitting DSP modules (see "DSP modules", page 53) and IP media modules (see "IP media module", page 60). The functions of the DSP chips on the DSP modules can also be configured.

System modules on the call manager card

The table below provides an overview of fixed DSP functions on the call manager card. Except for the Enterprise Voice Mail channels no licences or additional hardware are required to be able to use the functions.

Tab. 15 System modules on the call manager card

Max. number of simultaneous ...	Number of entries
Total circuits for the functions1) three-party conference, six-party conference, intrusion and silent intrusion2)	10
Circuits for the Call Waiting function	6
DTMF sender	9
DTMF receiver for voice mail or auto attendant	8
DTMF receiver for analogue terminals	8
Dialling tone receiver	2
Busy tone receiver	5
Ring receiver	2
FSK receiver 3) for CLIP detection on analogue network interfaces	4
CAS transmitter/receiver for PRI-E1 network interfaces4)	30
Total audio channels for basic voice mail5) or auto attendant2)	2
Total audio channels for Enterprise voice mail2), auto attendant2) or call recording2)	8

1) The functions can all be of the same type or used as a mix.

2) Licence required

3) One FSK transmitter available per FXS interface for CLIP display on analogue terminals. No media resources are required.

4) Of relevance only to certain countries such as Brazil

5) Can be used without licence subject to the following restrictions: Voice memory capacity approx. 20 minutes, no e-mail notification in the event of new voice messages, no forwarding of voice messages, no call recording, restricted voice mail menu by remote retrieval.

DSP function which can be selected on the call manager card

A DSP chip on the call manager card provides selectable functions. A description of the individual functions can be found as of page 54.

The functions are determined in the *Media resources* (*=ym*) view. In Tab. 20 all the possible combinations are listed, with the maximum number of channels. For this the DSP chip on the call manager card has to be loaded with different firmware. Additional functions require the use of one or more DSP modules. Some of these functions are subject to a licence.

3. 3 Expansion with cards and modules

An Mitel 470 basic system can be individually expanded using interface cards, system modules and an application card. The number and position of the available slots are described in the chapter "Interfaces, display and control elements", page 46.

3. 3. 1 System modules

With system modules a distinction is made between modules expandable as an option (DSP modules, IP media modules, Call charge modules) and mandatory modules (RAM module). This chapter describes only the system modules that can be expanded as an option. They expand the resources of the communications server, which means the system can be expanded step by step in line with requirements.

3. 3. 1. 1 DSP modules

Processor-intensive system functions require media resources. The communication server's DSP capacity increases through the use of DSP modules.

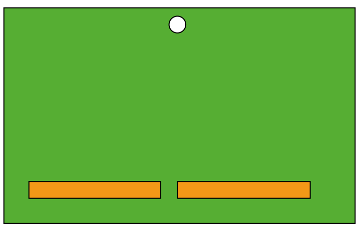


Fig. 12 Design of the DSP module

DSP modules are stacked on the call manager card and do not take up any slots for interface cards (see "Fitting DSP modules", page 112). The different types of modules can be used as a mix.

Tab. 16 DSP modules

Type	Number of DSP chips per module	Max. number of modules per system
SM-DSPX1	1	2
SM-DSPX2	2	
SM-DSP1[1)]	1	
SM-DSP2[1)]	2	

1) Although no longer available, the module is still supported.

Compared with DSP modules, modules with the designation DSPX are fitted with more powerful DSP chips. They are used to transmit VoIP data among others using the SRTP protocol (Secure VoIP).

Allocatable functions

One or more functions can be allocated to the individual DSP chips on the DSP modules and DSP chip on call manager card CPU1. For this the DSP chips have to be loaded with different firmware. The additional meadia resources can be used for DECT telephony, Voice over IP, fax transmissions, audio services, integrated mobile/external phones, additional dial tone and busy tone recipients on many analogue network interfaces FXO or for CAS (signalling protocol for PRI-E1 network interface in certain countries). This means that for each DSP chip a specific number of channels is available for the corresponding functions. Some of these functions are subject to a licence (see also "Licences", page 74).

Functions are allocated in WebAdminin the *Media resources* (*=ym*) view.

- *DECT*

  Operation of a DECT system on DSI interfaces with cordless phones. The voice data must be transformed on connections between DECT and non-DECT endpoints. This process requires DSP capacity.

  Purely DECT-DECT connections set up already do not require any media resources. On the other hand, media resources are required to set up connections.

  DECT channels can be used without a licence.
- *VoIP*

  Connections between IP and non-IP endpoints are made via an IP media gateway. This is carry out by the integrated standard media switch that switches VoIP channels for call connections in the IP network. The Standard Media Switch uses media resources for the real-time processing of the call data. VoIP channels are always required between IP and non-IP endpoints, e.g. for internal connections between an SIP/IP phone and a digital system phone or e.g. for an external user who is routed to the internal Voice Mail System via an SIP network interface. In an AIN VoIP channels are also used for call connections between the nodes (see "Use of VoIP channels", page 56 for an overview).

The number of configurable VoIP channels depends on both the type of DSP chip (see "Configuration of DSP chips", page 59) and the configured mode (see "Standard Media Switch modes of operation", page 58).

If the VoIP mode is set to G.711, two G.711 VoIP channels per system can be used without a licence. One *VoIP Channels for Standard Media Switch* licence is required for each additional VoIP channel.

> **Note**
> the IP media gateway function can also be provided with IP media modules. The necessary media resources are located on the IP media modules themselves. Standard media switch and IP media switch are independent of each other and can be used as a mix (see "IP media module", page 60).

- *FoIP*
  For reliable real-time fax transmissions via an IP network using the T.38 fax protocol (ITU-T). FoIP channels can be used without a licence.
- *Audio services*
  The audio channels are used to play back and record audio data. Additionally, each audio channel is assigned a DTMF receiver for enabling user inputs during playback. Licences (*Enterprise Voice Mail*, *Audio Record & Play Channels*, *Auto Attendant*) and media resources are required for this.

  Audio channels can be used for voice mail, auto attendant, queue with announcement, call recording, announcement with audio file, or conference bridge. The allocation is configurable (see "Reserving audio channels", page 58). Announcement service and music on hold use their own resources.

  The number of configurable audio channels depends on the type of DSP chip (see "Configuration of DSP chips", page 59).

  > **Note**
  > On the Mitel 470 communication server G.711 audio channels are always used for audio services. The *Voice mail mode* parameter can therefore not be changed for this system.
- *GSM*
  Enhanced functionality is achieved for integrated mobile/external phones by providing special DTMF receivers during the call connection. Suffix dialling functions (such as enquiry calls or setting up a conference with function codes) can be carried out as a result. The number of GSM channels – and therefore the number of DTMF receivers – depends on the number of users with integrated mobile/external phones who want to use this functionality simultaneously.

  One *Mobile or External Phone Extension* licence is required for each integrated mobile/external phone.
- *FXO*
  The basic resources (fixed DSP functions on the call manager card) cover 16 FXO

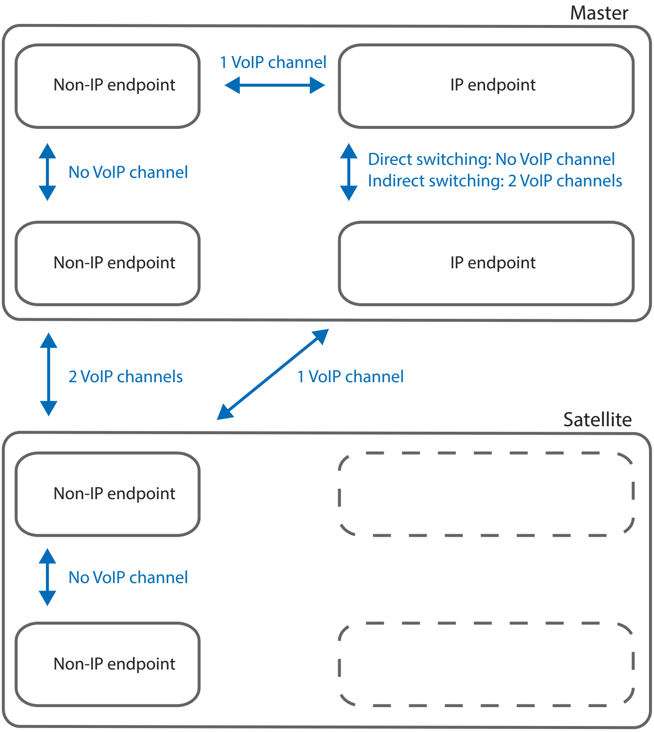
interfaces. For system configurations with more than 16 FXO interfaces this setting provides additional dialling tone and busy tone receivers.
Note: The values of the user-definable FXO channels corresponds to the number of FXO interfaces, not the number of additional dialling tone and busy tone receivers.

- *CAS*
  CAS (Channel-associated signaling) is a signalling protocol for PRI-E1 network interfaces used in certain countries (e.g. Brazil). Tone sender and receiver are required to transmit signalling information. Sufficient transmitters / receivers are already available for 1 PRI-E1 interface on the DSP of the call manager card (see Tab. 15). If this is not sufficient, additional senders/receivers can be configured with this setting.

Use of VoIP channels

VoIP channels are always required between IP and non-IP endpoints. They are freely available, i.e. they are always used wherever they happen to be needed. The figure below gives an overview of the cases where VoIP channels are needed and how many of them.

Tab. 17 Required VoIP channels between two possible endpoints



Non-IP endpoints:
- Analogue terminal (FXS)
- Digital system terminal (DSI)
- DECT cordless phone (DSI)
- ISDN phone (BRI-S)
- External via analogue exchange (FXO)
- External via ISDN exchange (BRI-T/PRI)
- Internal voice mail system
- Auto attendant
- Internal announcement service
- Music on hold
- Conversation recording
- Announcement with audio file
- Queue with announcement
- Conference bridge

IP endpoints
- IP system phone
- Mitel SIP terminal
- Standard SIP terminal
- DECT cordless phone via SIP-DECT
- WiFi cordless phone via SIP-DECT
- WiFi cordless phone via SIP access point
- WiFi mobile phone via MMC controller
- External via SIP provider

IP endpoints on satellites:
In normal operation all IP endpoints are registered with the master, even if they are located on the satellite.

Standard Media Switch modes of operation

The operation mode of the integrated standard media switch is defined with the *VoIP mode* parameter in the *Media resources* (*=ym*) view. The configured mode is always valid for the entire node.

Tab. 18 Integrated Standard Media Switch modes of operation

VoIP mode	Explanation	Licences
No VoIP	No VoIP channels can be configured.	
G.711	Although more voice channels are available per DSP in mode *G.711* than in hybrid mode, the volume of voice data is greater and requires a greater bandwidth.	Two VoIP channels per system can be used without a licence. One *VoIP Channels for Standard Media Switch* licence is required for each additional VoIP channel.
G.711/G.729	The VoIP hybrid mode *G.711/G.729* handles both G.711 and G.729 for coding voice data.	One *VoIP Channels for Standard Media Switch* licence is required for each VoIP channel.
Secure G.711	Same as *G.711* but with a more secure data transmission using the SRTP protocol.	One *VoIP Channels for Standard Media Switch* licence is required for each VoIP channel. The *Secure VoIP* licence, valid right across the system, is also required.
Secure G.711/G.729	Same as *G.711/G.729* but with a more secure data transmission using the SRTP protocol.	One *VoIP Channels for Standard Media Switch* licence is required for each VoIP channel. The *Secure VoIP* licence, valid right across the system, is also required.

Reserving audio channels

The allocation of audio channels between voice mail, auto attendant, call recording and announcements is set in the general voice mail settings (*=u1*).

An audio channel is always used for Auto attendant when an incoming call triggers greetings from mailboxes which are assigned an Auto Attendant profile. Audio channels of auto attendant are also used for queues with announcement. In all other cases one audio channel is used for voice mail in connection with the voice mail system.

Audio channels for call recording are used exclusively for the manual or automatic recording of phone calls.

Audio channels for announcements are used if the announcements have audio files. No audio channels are required for normal announcements by phone.

If no audio channel is reserved for any of the functions described above, or if all reserved audio channels are already used, audio channels from the *Non-reserved/shared* pool are used.

No audio channels can be reserved for conference bridges. Audio channels from the *Non-reserved/shared* pool are always used for the conference bridge.

Announcement service and music on hold use their own resources.

Tab. 19 Reserving audio channels

Parameter	Explanation
Available audio channels	Maximum available audio channels on this node. This value depends on the configuration of the media sources.
Reserved for Auto-Attendant	Number of audio channels on this node used for auto attendant and queue with announcement only.
Reserved for voice mail	Number of audio channels on this node that can be used exclusively for voice mail.
Reserved for call recording	Number of audio channels on this node that can be used exclusively for call recording.
Reserved for announcements	Number of audio channels on this node that can be used exclusively with audio file.
Non-reserved/shared	Number of audio channels on this node which can be used by voice mail, auto attendant, queue with announcement, call recording, announcement with audio file or conference bridge, depending on how they are needed at that time. Announcement service and music on hold use their own resources.

No audio channels are reserved after a first start and they can be used for voice mail, auto attendant, call recording or announcement.

Configuration of DSP chips

The functions which can be allocated to each DSP chip are determined in the *Media resources* (*=ym*) view. The DSP modules provide additional functions as indicated in the following table. All the possible combinations are listed, with the maximum number of channels.

Tab. 20 Max. number of channels per DSP chip on CPU1, SM-DSPX1 or SM-DSPX2

DECT	*VoIP* 1)	*FoIP*	*Audio* 1)	*GSM* 1)	*FXO*	*CAS* 2)	Remarks
10							
8			12				
8				5			
4			32	5			
4			24	10			
4			12	20			
4			12			150	
	5...8						Depends on the parameter *VoIP mode:* • *G.711*: 8 channels • *Secure G.711*: 7 channels • *G.711/G.729*: 6 channels • *Secure G.711/G.729*: 5 channels
	4		18	10			Only for *VoIP mode* = *G.711* or *G.711/G.729*
	4		12			150	Only for *VoIP mode* = *G.711* or *G.711/G.729*

DECT	*VoIP*[1]	*FoIP*	*Audio*[1]	*GSM*[1]	*FXO*	*CAS*[2]	**Remarks**
	3	3					
			46			150	
					64		

1) Licence(s) required (see also "Licences", page 74).

2) Of relevance only to certain countries such as Brazil

Tab. 21 Max. number of channels per DSP chip on SM-DSP1[1] or SM-DSP2[1]

DECT	*Audio*[1]	*GSM*[1]	**Remarks**
10			
8		10	
6	18	10	
	46		

1) Licence(s) required (see also "Licences", page 74).

Notes

- To configure VoIP channels on the DSP chip of a DSP module, make sure the *VoIP mode* parameter in the *Media resources* (*=ym*) view is not set to *No VoIP*. With the exception of the IP media modules the configured *VoIP mode* applies to all the DSP chips of a node. If *VoIP mode* is set to *G.711*, two G.711 VoIP channels per system can be used without a licence. The G.711 VoIP channels of the configurable DSP chip on processor card CPU1 can be combined with G.711 VoIP channels of DSP modules.
- If audio channels are configured and licensed, the two basic audio channels that can be used without a licence are redundant (see Tab. 15).
- Audio channels and FoIP channels can only be configured on one DSP chip per node.
- The system has to be restarted for the configuration changes of the DSP to take effect.
- After a first start all the DSP chips are configured on *DECT*.

3. 3. 1. 2 IP media module

IP media modules can be used for systems with high call switching requirements in the IP network. Depending on the module type a different number of VoIP and FoIP channels is available, provided by the IP media modules as required (see Tab. 23).

Note

The use of the IP media switch does not depend on the mode of operation of the standard media switch and the configuration of the DSP chips that are used by the standard media switch.

1) Although no longer available, the module is still supported.

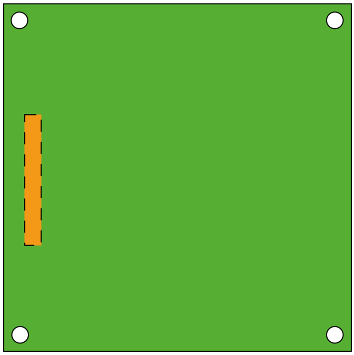


Fig. 13 Design of the IP media modules

IP media modules can be fitted both on the processor card CPU1 (see Fig. 9) and on the 1PRI/1PRI-T1 and 2PRI trunk cards (see Fig. 15). The modules are **not** stackable.

Tab. 22 IP media module

Type	Number of modules per CPU1 processor card	Number of modules per 1PRI/1PRI-T1[1)] trunk card	Number of modules per 2PRI trunk card	Max. number of modules per system
EIP1-8 EIP1-32[2)]	1	1	2	5

1) 1PRI not for USA/Canada, 1PRI-T1 only for USA/Canada.

2) The availability of this module depends on the sales channel.

The number of VoIP channels per IP media module depends on both the type of module and the use of voice channels:

Tab. 23 Max. number of voice channels per IP media module

Type	G.711 only, Secure G.711	G.711/G.729, Secure G.711/G.729	FoIP (T.38)
EIP1-8	32	8	8
EIP1-32[1)]	64	28	28

1) The availability of this module depends on the sales channel

3. 3. 1. 3 Call charge modules

Optional call charge modules are available for detecting charge pulses on analogue network interfaces.

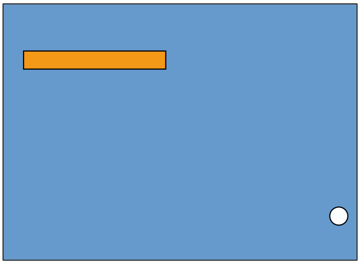


Fig. 14 Design of call charge modules

Call charge modules are fitted to FXO cards. The call charge modules available match the number of ports on the FXO cards. Only 1 call charge module can be fitted to each FXO card.

Tab. 24 Call charge modules

Type	Number of modules per 4FXO trunk card	Number of modules per 8FXO trunk card	Number of modules per 16FXO trunk card
4TAX[1)]	1	–	–
8TAX[1)]	–	1	–
16TAX[1)]	–	–	1

1) The availability of these modules depends on the sales channel

3. 3. 2 Interface cards

Interface cards are fitted from the front into one of a total of 7 expansion slots (see "Fitting interface cards", page 110). Interface cards can be assigned to two categories:

- Trunk cards
  These cards provide interfaces for connection to public dial-up networks or for networking systems to create a private telephony network.
- Terminal cards
  These cards provide interfaces for connecting digital and analogue voice and data terminals.

On some BRI cards a part of the interfaces are configurable (BRI-S/T). This means that these cards cannot be clearly assigned to any particular category. They are listed both among the trunk cards and the terminal cards.

Up to 2 IP media modules can be fitted on PRI cards.

Each FXO card can be fitted with one call charge module.

The number of RJ45 sockets on the front depends on the type of interface card. On cards with 16 or more interfaces part or all of the RJ45 sockets are multiply assigned.

They are fed to the fan-out-panel (FOP) using patch cables and then split to individually assigned RJ45 sockets (see "Fan-out panel FOP", page 157).

The splits can also be made elsewhere, e.g. using system cables available separately (see "Prefabricated system cable 4 x RJ45", page 117).

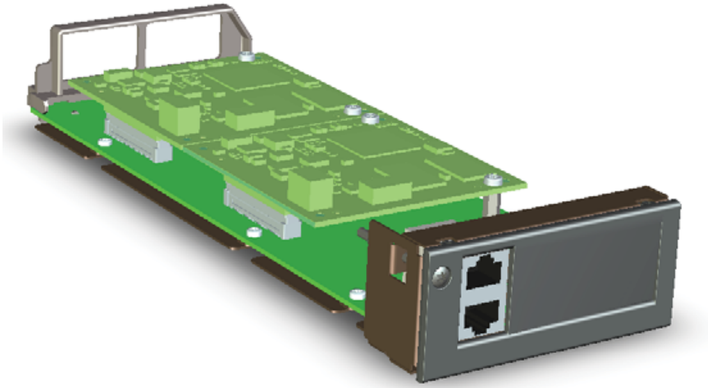


Fig. 15 Example of an interface card (2PRI with 2 IP media modules fitted)

3. 3. 2. 1 Trunk cards

The trunk cards contain interfaces for connection to the analogue public network (PSTN), the digital public network (ISDN) or for networking systems to create a private telephony network (PISN). The trunk cards can be used and operated on any slots for interface cards.

The trunk cards contain either FXO interfaces (FXO: Foreign Exchange Office), PRI interfaces (PRI: Primary Rate Interface) or BRI interfaces (BRI: Basic Rate Interface).

BRI cards contain both network interfaces (BRI-T) and terminal interfaces (BRI-S). On the BRI cards 4 interfaces can be individually configured to BRI-S or BRI-T.

Tab. 25 Trunk cards

Type	Network interfaces per card	Max. number of cards per system	Remarks
1PRI[1]	1 × PRI-E1	7[2]	• Can be fitted with 1 IP media module • Contains 30 B channels • 10 B channels can be used licence-free • **Not** usable in USA/Canada for the public network
1PRI-T1[1]	1 × PRI-T1	7[2]	• Can be fitted with 1 IP media module • Contains 23 B channels • 10 B channels can be used licence-free • **Only** usable in USA/Canada for the public network
2PRI[1]	2 × PRI-E1	7[2]	• Can be fitted with 2 IP media modules • Contains 2 × 30 B channels • 2 ×10 B channels can be used licence-free • **Not** usable in USA/Canada for the public network
4BRI[1]	4 × BRI-T	7[2]	• All interfaces configurable to BRI-S • **Not** usable in USA/Canada for the public network
8BRI[1]	8 × BRI-T	7[2]	• Four fixed BRI-T interfaces • 4 BRI-T interfaces configurable to BRI-S • **Not** usable in USA/Canada for the public network
4FXO[1]	4 × FXO	7[2]	• 1 call charge module can be fitted for 4 ports
8FXO[1]	8 × FXO	7[2]	• 1 call charge module can be fitted for 8 ports
16FXO[1]	16 × FXO	4	• 1 call charge module can be fitted for 16 ports

1) The availability of these cards depends on the sales channel

2) 1 fewer card if CPU2 application card is fitted

3. 3. 2. 2 Terminal cards

Terminal cards are used for connecting digital and analogue voice and data terminals.

FXS cards are an exception. Their analogue interfaces are multifunctional. In addition they provide interfaces for controlling external devices and switching over internal switch groups. Depending on the terminal or function, the interfaces are configured individually and switched over internally accordingly (see "Multifunctional FXS interfaces", page 151).

DSI cards are used for connecting digital system terminals such as phones. 2 terminals can be connected to each DSI interface.

Terminals to ETSI standard are connected via BRI cards. The cards contain both terminal interfaces (BRI-S) and network interfaces (BRI-T). On the BRI cards 4 interfaces can be individually configured to BRI-S or BRI-T.

Tab. 26 Terminal cards

Type	Terminal interfaces per card	Max. number of cards per system	Remarks
4FXS	4 × FXS	7[1]	• Interfaces individually configurable • 2 interfaces on each card (X.1 and X.2) are designed for long lines.
8FXS	8 × FXS	7[1]	• Interfaces individually configurable • 2 interfaces on each card (X.1 and X.2) are designed for long lines.
16FXS	16 × FXS	7[1]	• Interfaces individually configurable • 2 interfaces on each card (X.1 and X.2) are designed for long lines. **Note:** To prevent the system from overheating, no more than 50 FXS ports should be active simultaneously on each system.
32FXS	32 × FXS	7[1]	• Interfaces individually configurable • 2 interfaces on each card (X.1 and X.2) are designed for long lines. **Note:** To prevent the system from overheating, no more than 30% of the FXS ports should be active simultaneously per 32FXS card and no more than 50 FXS ports per system.
8DSI[2]	8 × DSI	7[1]	
16DSI[2]	16 × DSI	7[1]	
32DSI[2]	32 × DSI	7[1]	
4BRI[2]	4 × BRI-S	7[1]	• All interfaces configurable to BRI-T • **Not** usable in USA/Canada for the public network
8BRI[2]	4 × BRI-S	7[1]	• Four fixed BRI-T interfaces • 4 BRI-S interfaces configurable to BRI-T • **Not** usable in USA/Canada for the public network

1) 1 fewer card if CPU2 application card is fitted

2) The availability of these cards depends on the sales channel

3. 3. 3 Applications card CPU2-S

The applications card is connected with the call manager call via Ethernet and the backplane, which means that the Ethernet interface on the front panel is not required.

The Mitel Mitel Open Interfaces Platform (OIP) applications and a fax service are already pre-installed on the application card standard PC.

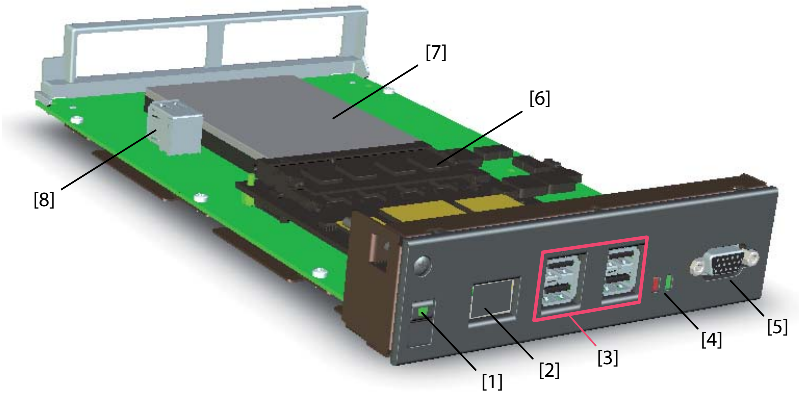


Fig. 16 Interfaces, display and control elements of the applications card

Tab. 27 Interfaces, display and control elements of the applications card

Interfaces, display and control elements	Number of entries	Position	Remarks
On/Off button with integrated status LED	1	[1]	
Ethernet interfaces: 1Gbit/s	1	[2]	No provision for use at present
USB interfaces 2.0	4	[3]	For connecting the keyboard, mouse, etc.
Status LEDs	2	[4]	For indicating HDD access and USB supply overload
VGA video interface	1	[5]	For connecting the monitor
Processor module with standard PC	1	[6]	
> 250 GB hard disk	1	[7]	
USB interfaces 2.0 for "software dongles"	2	[8]	

The meaning of the status LEDs is explained in the chapter "Application server display and control panel", page 222.

The maximum permissible current input at the USB interfaces varies:

Tab. 28 Max. admissible current input at USB interfaces

Front-side USB interfaces	Internal USB interfaces	Max. current input [mA]
top left / bottom left	bottom	100
top right / bottom right	top	500

Access to the applications server is normally via the WebAdmin configuration tool, which means the front-side interfaces of the applications card are not needed.

Note

For licensing reasons the front-side connections are to be used for maintenance purposes only. Installing user-specific applications is prohibited.

See also:
For more information about installing, configuring and upgrading the software of the application card, see the CPU2-S application card installation manual.

3. 4 System capacity

System capacities are defined on the one hand by the existing hardware with its expansion possibilities and on the other by the limits set in the software. The software limits can be partly expandable by licences.

3. 4. 1 Media resources

3. 4. 2 General system capacity

The number of slots, interface cards and system modules per communication server have already been mentioned in the previous chapters and are not listed separately in this chapter.

Tab. 29 General system capacity

Max. number...	Mitel 470	AIN with Mitel 470 as Master
Nodes in a transparent network (AIN)	–	41
Nodes with SIP networking	100	100
Users	600[1)2)]	600
Terminals per user[3)]	16	16
Simultaneous connections		
• Without IP and without DECT (internal / external)	184	250
• IP – not IP (internal / external)	184	250
• IP – IP (internal)	250	250
• IP – IP via SIP access channels (external)	240	240
• DECT – not DECT (internal / external)	50	250
• DECT – DECT (internal)	184[4)]	250
• MiCollab - connections	80	80
Voice channels VoIP G.711 / G.729 (Standard Media Switch)[5)]	24 / 24	500
Voice channels VoIP G.711 / G.729 (IP media switch)[6)]	250 / 140	250 / 250
Audio channels, call recording	8	per node[7)]
Audio channels for voice mail	16	per node
Audio channels for voice mail and call recording, total	16	per node
Audio channels for auto attendant	46	per node
Total audio channels[8)]	46	per node
Voice channels FoIP, T.38 (standard media switch)	3	per node

Max. number...	Mitel 470	AIN with Mitel 470 as Master
Voice channels FoIP, T.38 (IP media switch)	140	per node
CAS transmitter/receiver for PRI-E1 network interfaces[9)]	150	per node
Configurable conference bridge	60	60
Active conferences	see Tab. 15	
Trunk group	506	506
Trunk groups in route	8	8
Network interfaces per trunk group	64	64
Routes	212[10)]	212[10)]
B channel groups	506	506
SIP provider	10	10
SIP user account	1200	1200
Direct dialling plans	10	10
Total DDI numbers[11)]	4000	4000
SmartDDI conversion rules per DDI plan	100	100
SmartDDI conversion rules overall	200	200
Call distribution elements	4000	4000
Queue with announcement	16	16
User groups	99	99
Members per user group "normal"	16	16
Members per user group "large"	400	600
Abbreviated dialling numbers + PISN users	4000	4000
Operator keys per phone on Mitel 6800/6900 SIP	10[12)]	10[12)]
Room keys on Mitel 6873 SIP (inclusive expansion keypad)	200	200
Line keys per key telephone (except Mitel 6800/6900 SIP)	39	39
Line keys per key telephone on Mitel 6800/6900 SIP	2...12[13)]	2...12[13)]
Line keys per CDE on Mitel 6800/6900 SIP	16[14)]	16[14)]
Total line keys on Mitel 6800/6900 SIP	see [15)]	see [15)]
Switch groups	50	50
Positions per switch group	3	3
Hotline destinations	20	20
Emergency destinations	50	50
Internal emergency numbers	10	10
Internal emergency response teams	20	20
Members of internal emergency response teams	20	20
Public emergency numbers	20	20
Allocations of external call numbers to internal call numbers	1000	1500
External digit barring	16	16
Internal digit barring	16	16

Max. number...	Mitel 470	AIN with Mitel 470 as Master
Predefined text messages	16	16
Announcement / message groups	50	50
User per announcement / message group	16	16
Data service tables	32	32
User accounts for user access control	25	25
Authorization profiles for user accounts	25	25
Log entries per user account	20	20
First-party CTI users via LAN	32	32
First-party CTI users via Mitel Dialer	600	600
Third-party CTI interfaces	1	1
Third-Party CTI interface (Basic, Standard)	600	600
Groups, Agents (OIP Call centre)	150	150
Agents (MiContact Center Business)	80[16)]	80[16)]
Mailboxes with Basic or Enterprise voice mail system	600	600
Greetings per mailbox	3	3
Profiles per mailbox for auto attendant	3	3
Backup communication servers for Dual Homing	50	50
Primary communication servers for Dual Homing	50	50
Blacklist	1	1
Call number entries in the blacklist	3000	3000
Number of CLIP based routing tables	20	20
Total call number entries in call distribution tables	1000	1000
Call data memory internal (number of records)[17)]	1000	1000
Private contacts	12000	12000
Call list entries for each of the 3 call lists per phone	30	30
Total call list entries	60000	60000
Busy lamp field keys on Mitel SIP phones in total	4000	4000
Busy lamp field keys per Mitel SIP phone	50	50
Same users on busy lamp field keys on Mitel SIP phones	25	25
Configured keys	48000	48000
Expansion key modules on DSI terminals	400	400
Expansion key modules on IP system phones	400	400
Expansion key modules on Mitel 6800/6900 SIP phones	600	600
Alpha keyboard Mitel K680	400	600
Alpha keyboard (AKB)	400	400

1) Each user requires a licence.

2) For Russia maximum 256 users

3) Only 1 operator console, 1 MiVoice 2380 IP, 1 BluStar 8000i , 1 Mitel BluStar for PC, 1 Mitel SIP-DECT, 2 DECT-cordless phones and 1 MiCollab client (3 MiCollab clients with MiCollab version 8.1) are possible for each user.

4) This is the maximum value for connections set up already. Since media resources are required to set up connections, this value may be reduced.

5) In the Secure VoIP modes the maximum values cannot be achieved with the selection i the DSP settings: *Secure G.711* VoIP mode: 3 ´ 7 = 21 channels, VoIP mode *Secure G.711/G.729*: 4 × 5 = 20 channels

6) Applies also to Secure VoIP modes

7) For IP-IP connections maximum 8

8) Audio channels can be used for voice mail, auto attendant, queue with announcement, call recording, announcement with audio file, or conference bridge. Announcement service and music on hold use their own resources.

9) Of relevance only to certain countries such as Brazil

10)12 of them are masked (not configurable)

11)In USA/Canada the abbreviation DID (Direct Inward Dial) is used instead of DDI (Direct Dialling In)

12)Only 6 on Mitel 6940 SIP/Mitel 6873 SIP if phone is also used as reception phone.

13)Depending on the phone type: Aastra 6730i/31i: 6 keys; Mitel 6735/37/39/53/55/57 SIP: 9 keys; Mitel 6863 SIP: 2 keys; Mitel 6865/67 SIP: 9 keys; Mitel 6869/73 SIP: 12 keys; Mitel 6900 SIP: 12 keys

14)The value applies to CDE with destination KT line. With multiple destinations (User + KT or KT + UG) the value is reduced to 8.

15)Depending on the highest number of line keys, configured for the same line. The following pairs apply (line keys per line / total line keys): (16/48), (14/56), (12/72), (10/100), (8/160), (6/240), (4/320), (2/400).
Example: The following line keys are configured on different Mitel SIP phones: 8 keys for line 1, 14 keys for line 2, 10 keys for line 3, 10 keys for line 4.
→ Highest number of keys per line: 14
→ total 56 line keys are allowed
→ Configured line keys: 8 + 14 + 10 + 10 = 42 → OK

16)Only 56 with analogue network interfaces

17)The call data memory is only used if the output destination is blocked (e.g. printer jam).

Tab. 30 System capacity application card CPU2-S

Max. number...	CPU2-S
Fax server: Fax mail boxes / media channels	600 / 8
Mitel 400 Call Center: Agents / groups	50 /50
Mitel 400 CCS: Supervisors / wallboards	20/20
Mitel OfficeSuite users	200
MiVoice 1560 users	5
Integration of phone directories	5
Constant load (calls per hour)	1000

3. 4. 3 Terminals

Tab. 31 Maximum number of terminals per system and interface

Interface	Terminal type	Terminal	per Mitel 470	per AIN with Mitel 470 as Master	per interface
Miscellaneous	Terminals (including virtual terminals and integrated mobile/external phones)		600	600	
Miscellaneous	Terminals (excluding virtual terminals and integrated mobile/external phones)		600	600	
Miscellaneous	Free seating pools		600	600	
DSI-AD2	Terminals on DSI-AD2 interfaces (total)		448	600	
DSI-AD2	Digital system phones	MiVoice 5360 MiVoice 5361 MiVoice 5370 MiVoice 5380	448	600	2
DSI-AD2	Operator consoles / operator applications	MiVoice 5380 MiVoice 1560	32	32	2
DSI-AD2	Cordless system	SB-4+ radio unit	224[1)]	255[1)]	1
DSI-AD2	Cordless System	SB-8 / SB-8ANT radio units	112[1)]	255[1)]	[2)]
DSI-DASL	Digital system phones	Dialog 4220 Dialog 4222 Dialog 4223	224	600	1
DECT	Cordless phones	Mitel 610/612 DECT Mitel 620/622 DECT Mitel 630/632 DECT Mitel 650 DECT Office 135 Office 160 GAP terminals	600	600	
LAN	Terminals on LAN interfaces (total)		600	600	
LAN	DHCP clients on the internal DHCP server		400	400	
LAN	IP terminals	MiVoice 2380 IP MiVoice 5360 IP MiVoice 5361 IP MiVoice 5370 IP MiVoice 5380 IP	600	600	
LAN	IP operator consoles / IP operator applications	Mitel 6930 SIP Mitel 6940 SIP Mitel 6869 SIP Mitel 6873 SIP	4	4	
		MiVoice 5380 IP MiVoice 1560	32	32	
LAN	Reception/Front desk	Mitel 6940 SIP Mitel 6873 SIP	4	4	

Interface	Terminal type	Terminal	per Mitel 470	per AIN with Mitel 470 as Master	per inter-face
LAN	Mitel SIP terminals	Mitel 6920 SIP Mitel 6930 SIP Mitel 6940 SIP Mitel 6863 SIP Mitel 6865 SIP Mitel 6867 SIP Mitel 6869 SIP Mitel 6873 SIP	600	600	
LAN	Mitel SIP-DECT Cordless phones		600	600	
LAN	Standard SIP terminals		600	600	
LAN	Mitel BluStar 8000i		600	600	
LAN	Mitel BluStar Softphones		600	600	
LAN	Mitel Mobile Client Controller		10	10	
–	Virtual terminals		600	600	
–	Integrated mobile phones without MMC		255	255	
–	Integrated mobile phones with MMC		600	600	
–	Integrated mobile phones per MMCC Compact		50	50	
–	Integrated mobile phones per MMCC 130		250	250	
–	Integrated external phones (e. g. for Skype for Business)		600	600	
BRI-S	Terminals on BRI-S interfaces (total)		224	512	8[3)]
BRI-S	Terminals as per ETSI standard • ISDN terminals • ISDN PC cards • ISDN LAN routers • ISDN Terminal Adapters		224	512	
FXS	Terminals on FXS interfaces (total)		228	600	1
FXS	Analogue, nationally approved terminals • Pulse dialling (PUL) • Frequency dialling (DTMF) • Radio units for cordless phones • Door intercoms with DTMF control functions • Group 3 fax machines[4)] • Answering machines • Modems		228	600	
FXS	External audio equipment with line output		1	1 per node	
FXS	External equipment can be switched via control outputs		228	600	
FXS	External switches for controlling internal switch groups via control inputs		228	600	
FXS	General bell		1	1 per node	

1) Maximum 64 radio units per location area if 4 location areas are defined, or maximum 128 radio units per location area if 2 location areas are defined.

2) Operation on 2 DSI interfaces in each case

3) Maximum of 2 simultaneous call connections.

4) Transmission with the T.38 protocol is recommended for Fax over IP. The corresponding media resources need to be allocated.

3. 4. 4 Terminal and network interfaces

Tab. 32 Terminal and network interfaces

Max. number...	Mitel 470	AIN with Mitel 470 as Master
Ethernet interfaces	3	per node
Network interfaces, total (FXO, BRI-T, PRI, BRI-Sext.)	56	288
Terminal interfaces, total (DSI, FXS, BRI-S)	228	600
DSI terminal interfaces	224	600
Analogue terminal interfaces FXS	228	600
BRI-S terminal interfaces	28	224
Analogue network interfaces FXO	64	64
Basic rate interfaces BRI-T	56	256
Basic accesses BRI-S ext.	28	256
Primary rate interfaces PRI[1)]	14	32
SIP access	10	10
SIP access channels[2)]	240	240

1) 10 B channels per PRI network interface can be used without licence

2) Licences required

3. 4. 5 Software assurance

Software Assurance (SWA) is Mitel's comprehensive support offer which gives access to new software releases, support services and SRM remote access to the communication server.

The software assurance agreement has a fixed runtime and defines the number of authorised users on the communication system. You can see at a glance whether a valid (active) SWA is available for the communication server, via the SWA state in the WebAdmin title bar.

The SWA state is retrieved via an encrypted direct link on the licence server. If there is no connection to the licence server, the last known state is displayed

The number of users covered via SWA and the number of configured users requiring SWA can be seen in the *System information* (=1v) view. SWA becomes invalid if the number of configured users exceeds the number of users covered via SWA.

3. 4. 6 Licences

Use of the call manager software requires a licence. Additional licences are required in order to use a number of enhanced functions and protocols, to enable voice channels or to operate certain terminals. The Mitel CPQ application automatically plans the necessary licences, which are then enabled on the communication server using a licence file.

The licence file contains all the enabled licences. When you purchase a new licence from your authorised dealer, you obtain a new licence file in return. Upload this file in WebAdmin in the *Licences* (*=q9*) view.

Notes:
- A licence file is not transferable to another communication server.
- If you receive a voucher instead of a licence file, log on with your partner login at Mitel Connect https://connect.mitel.comand generate the licence file yourself using the EID number. Detailed instructions about this can be found in the WebAdmin help on the *Licences* (*=q9*) view.

3. 4. 6. 1 Description of available licences

Software

- *Software Release*

  Updating to a new software release requires a licence. A valid software assurance (SWA) entitles you to upgrade the communication server to a new software level for a specific period., and to operate it with a specific number of users.

  A valid software assurance is the prerequisite for being able to acquire an update licence (*Software Release* licence) for a particular software version. Without a valid *Software Release* licence you can update the communication server to a new software level, but after four hours of operating time it will switch over to the restricted operating mode (see "Restricted operating mode", page 83). The communication server will switch back to normal operation as soon as you upload a licence file that comprises the *Software Release* licence. You do not need to restart the communication server.

  Note:
  The purchase of a new communication server also includes a software assurance for a specific period. Log on with your partner login to Mitel Connect https://connect.mitel.com and obtain a new licence file using the EID number and the voucher. The licence file issued as a result contains the appropriate *Software Release* licence (and any other licences you may have acquired). You can now activate the communication system with this licence file. Detailed instructions about this can be found in the WebAdmin help on the *Licences* (*=q9*) view.

Mitel Advanced Intelligent Network

In an AIN, a valid *Software Release* licence must be available on the master only. Exception: For long-term offline mode, for operation with Secure VoIP and use as backup communication server, the satellite must also have a valid *Software Release* licence.

- Behaviour of satellites in online mode:
  Although satellites must also have a release licence, they must not necessarily match the current software status. If satellites do not have any release licence, they restart every four hours.
- Behaviour of satellites in offline mode:
  Satellites with an incorrect release licence change over to restricted operating mode after thirty-six hours. Satellites without any release licence change over to restricted operating mode after four hours.

Users

- *User*
  Mitel 470 requires a *User* licence for each user in the system.
  Exception: A user without a terminal or with a virtual terminal only does not need a licence.
  Note: The *Mitel 470 Base licence* (see page 79) contains already *User* licences.
- *Basic User* (licence bundle)
  With this licence bundle an additional user is available who can assign any type of terminal including the appropriate phone licence, if needed. This allows the user to change the phone type without changing the licensing. Note that with this licence bundle only one terminal can be assigned to an user. The licence bundle is explicitly assigned to an certain user.
- With the following UCC licence bundles an additional user is available who can assign 8 terminals of any type including the appropriate phone licences and video licences for all phones, if needed. The licence bundles are explicitly assigned to an certain user:
  - *Entry UCC User*
    This licence bundle contains the licences described in the above section and activates MiCollab functions for the MiCollab role *UCC Entry*.
  - *Standard UCC User*
    This licence bundle contains the licences described in the above section and activates MiCollab functions for the MiCollab role *UCC Standard*.
  - *Premium UCC User*
    This licence bundle contains the licences described in the above section and activates MiCollab functions for the MiCollab role *UCC Premium*.

  With a specific number of UCC licence bundles, users with SIP terminal licences for using with MiCollab AWV are added.
  The formula is: **10 + [Standard UCC User] / 10 + [Premium UCC User] / 5**

Example: Entry UCC User: 12, Standard UCC User: 22, Premium UCC User: 14
Formula: 10 + 22 / 10 + 14 / 5 = 14 users with SIP terminals.

With a specific number of UCC licence bundles more voice mail channels licences are added.
The formula is: **([UCC licence bundles of any type] - 10) / 10**
Example: Entry UCC User: 12, Standard UCC User: 22, Premium UCC User: 14
Formula: UCC licence bundles: 48: (48 - 10) / 10 = 3 additional voice mail channels

Terminals

- *MiVoice 2380 IP Softphones*
  One licence per terminal is required to operate the IP softphones MiVoice 2380 IP. The licences are needed to register the terminals on the system.
- *MiVoice 5300 IP Phones*
  One licence per terminal is required to operate the IP system phones MiVoice 5360 IP, MiVoice 5361 IP, MiVoice 5370 IP and MiVoice 5380 IP. The licences are needed to register the terminals on the system. If the required licences are missing, the relevant event message is output on the system. The licences can also be used if the *Mitel SIP Terminals* licences are missing (but not the other way round).
- *Mitel SIP Terminals*
  To operate Mitel SIP terminals of the Mitel 6800/6900 SIP series, for cordless terminals logged on via Mitel SIP-DECT or Mitel SIP WLAN base stations, one licence is required per terminal or user. The licences are needed when registering the terminals or the user on the system. If the licences are missing, Mitel SIP terminals can also be operated with *SIP Terminals* or *MiVoice 5300 IP Phones* licences (but not the other way round).
- *Mitel Dialog 4200 Phones*
  One licence per phone is required to operate Dialog 4220, Dialog 4222 and Dialog 4223 digital phones. The licences are needed to register the phones on the system.
- *MMC Extension*
  With this licence mobile phones can be integrated into the communication system together with an Mitel Mobile Client Controller and Mitel Mobile Client. The MMC Controller allows mobile users to move back and forth between the internal WLAN coverage and the mobile radio network without the call being interrupted.
- *Dual Homing*
  In the event of failure of the primary communication server or an interruption in the IP connection to the primary communication server, SIP phones in the Mitel 6800/6900 SIP series can automatically register on a backup communication server. On the **backup communication server one licence** is required per phone. The licences are needed to register the clients on the system.

- *Mobile or External Phone Extension*
  With this licence it is possible to integrate mobile phones or other external phones into the communication system. One licence has to be purchased for each phone.

  Note:
  This licence does **not** allow comfortable integration with the Mitel Mobile Client application.
- *SIP Terminals*
  One licence is required per terminal to operate standard SIP terminals. The licences are needed when registering the terminals on the system and can be used even if *Mitel SIP Terminals* licences are missing (but not the other way round).
- *Video Terminals*
  To be able to use the video functionality of a standard SIP video terminal you need to acquire a Video Terminals licence in addition to a *SIP Terminals* licence. The licences can also be used if the *Mitel 8000i Video Options* licences are missing.

BluStar

- *BluStar Softphones*
  This is a BluStar client licence. One licence per client is required to operate BluStar softphones. The licences are needed to register the clients on the system.
- *BluStar Softphone Video Options*
  This licence is required for using the video functionality of a BluStar softphone. A BluStar client licence must be in place.

Audio services

- *Conference Bridge* (Dial-In conference)
  This licence allows the use of a conference bridge. The internal or external conference participants choose a specific call number and are connected with the conference after entering a PIN. One licence is required per system /AIN.
- *Number in Queue*
  This licence is required for using the functionality of "Queue with announcement". The *Auto Attendant* licence is required here. One licence is required per system /AIN.
- *Auto Attendant*
  This licence enables the use of the auto attendant function and is independent of the Enterprise Voice Mail licence. It means it can also be used in conjunction with basic voice mail. One licence is required per system /AIN.

  Note
  In a VoIP environment VoIP channel licences are also required for converting the voice data when using the auto attendant.

- *Enterprise Voice Mail*
  If the functionality of the basic voice mail system is insufficient, the voice mail system can be expanded. This licence provides 2 audio channels for recording or playing back audio data for voice mail, auto attendant or call recording. The licence also increases the voice memory capacity and allows e-mail notification whenever new voice messages are received as well as the forwarding of voice messages and call recording.

  Notes
  – Additional audio channels require additional *Audio Record & Play Channels* licences. An *Auto Attendant* licence is required to use the auto attendant function.
  – In a VoIP environment VoIP channel licences are also required for converting the voice data when using the internal voice mail system.

- *Audio Record & Play Channels*
  This licence enables an additional audio channel for recording or playing back audio data for voice mail, auto attendant or call recording. This licence can only be used in conjunction with the *Enterprise Voice Mail* licence.

  Mitel Advanced Intelligent Network
  In an AIN the Enterprise Voice Mail and Audio Record & Play Channels licences are all acquired for the Master. The number of Audio Record & Play Channels licences determines the maximum number of simultaneously active audio channels, regardless of the nodes on which they are currently being used. Requirement: The media resources on each node must be available and allocated accordingly.

Features

- *Analogue Modem*
  This licence allows remote maintenance of an Mitel 415/430 using an analogue modem. For this the *Modem* function must be allocated to the mainboard DSP. Transmitting event messages via an analogue modem is also possible.

  Mitel Advanced Intelligent Network
  In an AIN the licence is always acquired on the Master. The licence allows the remote maintenance of the AIN via any Mitel 415/430 node.
  Note: The master node can also be of Mitel SMBC, Mitel 470 or Virtual Appliance type.

- *Secure VoIP*
  This licence allows encrypted VoIP connections with the aid of SRTP (Secure Real-Time Transport Protocol) and/or encrypted SIP signalling data using TLS (Transport Layer Security).

  Mitel Advanced Intelligent Network
  For legal reasons (Trade Control Compliance) in an AIN a *Secure VoIP* licence is required for both the Master and for each satellite.

- *Silent Intrusion*
  This licence is needed for the *Silent intrusion* feature, which is similar to the *Intru-*

sion feature. The difference is that the user intruded upon receives neither a visual nor an acoustic signal of the intrusion. The feature is used mainly in call centres. One licence is required per system /AIN.

Resources

- *Mitel 470 Base licence* This basic licence is required for Mitel 470. It contains 20 *User* licences (see page 75). With this basic licence no other licences are needed for setting up a Mitel Advanced Intelligent Network (AIN).
- *VoIP Channels for Standard Media Switch*

Note:
This licence is required for Mitel 415/430, Mitel SMBC and Mitel 470 only. For Virtual Appliance, the VoIP channels of the integrated Mitel Media Server are made available and do not require any licences.

This licence enables the conversion of voice channels for VoIP-non-VoIP connections and is used for IP terminals, SIP terminals, SIP access channels or to operate an Mitel Advanced Intelligent Network. High voice data compression is possible with the G.729 VoIP channels. An additional voice channel is activated with each licence.

Notes:
- If VoIP mode is set to G.711, two G.711 VoIP channels per system can be used without a licence.
- Theoretically there are no VoIP channel licences in a pure VoIP environment (only IP/SIP phones on the system and connection to the public network via an SIP provider). However, as soon as voice mail functions, the announcement service or music on hold is used, VoIP channel licences are required as the use of these functions entails a conversion of the voice data.

Mitel Advanced Intelligent Network
In an AIN the licence can also be used for the connections between the nodes. Two VoIP channel licences are required for each node connection. The licences are always acquired for the Master. The number of licences determines the maximum number of simultaneously active conversions, regardless of the nodes on which they are currently being used.
Requirement: The media resources on each node must be available and allocated accordingly.
If Virtual Appliance is used as Master, the VoIP channels of the master node are made available without a licence from the integrated Mitel Media Server. However, for the satellites' VoIP channels, the licences must be purchased.

Networking

- *Lync Option for SIP Access Channels*
  This additional licence enables the use of a SIP access channel with Lync-specific options and features. It is required for each channel in addition to a *SIP Access Channels* licence.

- *B-Channels on PRI Cards*
  10 B-channels can be used without licences for each PRI interface. These channels cannot be transferred to other PRI interfaces. An additional channel is activated with each licence. These licences are in a pool and are used from any PRI interface, if necessary (per call).

  Mitel Advanced Intelligent Network
  In an AIN the licence is always acquired on the Master. For each licence an additional B channel is available on a PRI interface of any node, depending on where the B channel is currently being used.

- *SIP Access Channels*
  The connection of the system to a SIP service provider or the networking of the systems via SIP requires one licence per channel.

  Mitel Advanced Intelligent Network
  In an AIN all the SIP licences are always acquired for the Master. The number of licences determines the maximum number of simultaneously active voice channels, regardless of the nodes on which they are currently being used. Requirement: The media resources on each node must be available and allocated accordingly.

Private networking

QSIG Networking Channels
These licences are used to implement a private leased-line network with QSIG by enabling a specific number of simultaneously outgoing QSIG channels. Two licence levels are available (see Tab. 33).
Note: For Virtual Appliance this licence is only relevant to the QSIG networking of an AIN satellite.

Applications

- *Advanced Messaging*
  Enables the SMPP protocol to be used for integrating an SMS server and 9d cordless phones to be logged on as system phones (Ascom Wireless Solutions products). User-friendly messaging systems can then be implemented. One licence is required per system/AIN.
- *CTI First Party via LAN*
  This basic licence enables the CTI basic functions via Ethernet interface (e.g. for using a PC dial help) for a specific number of users (see "General system capacity", page 67). It cannot be combined with CTI third-party licences.
- *Dialers*
  This licence allows you to use the Mitel Dialer CTI application. The number of licences determines the simultaneously active, user-assigned Mitel Dialer applications.

- Licences for the fax service on the CPU2
  The CPU2 applications card of an Mitel 470 communication server contains software with a server-based fax solution. Use of this fax service is licensed as follows:
  - *CPU2 Fax Base*
    This licence comprises 2 *CPU2 Fax Channels* and 10 *CPU2 Fax Clients* licences. This means that 2 fax messages can be sent or received simultaneously and 10 users can be assigned a fax mailbox.
  - *CPU2 Fax Channels*
    Additional media channels for simultaneously transmitting and receiving fax messages (maximum number = 8 media channels).
  - *CPU2 Fax Clients*
    Additional users configurable with fax mailbox.
- *Hospitality Manager*
  This licence allows you to use the Mitel 400 Hospitality Manager. The Mitel 400 Hospitality Manager is a web-based application for receptionists in the hospitality sector. One licence is required per system /AIN.
- *Hospitality PMS Interface* and *Hospitality PMS Rooms*
  The *Hospitality PMS Interface* licence is used to connect the communication server to a hotel management system using the FIAS protocol. One licence is required per system /AIN. Moreover, one *Hospitality PMS Rooms* licence is required per room.
- OpenCount licences
  MitelOpenCount is a software package used for the call logging management on the communication system. It consists for selected sectors of basic, comfort and premium solutions and is installed on an external server. The licences are stored in MiVoice Office 400. OpenCount obtains the licences via the XML based interface Open Application Interface.
  - *Mitel OpenCount Basic Package*
    This basic licence is a prerequisite for all OpenCount additional licences. The licence contains the “Company” branch package, enables the connection to MiVoice Office 400 and allows basic functions to be used.
  - *Mitel OpenCount Healthcare Branch Package*
    This additional licence offers extra functions for care and retirement homes.
  - *Mitel OpenCount Public Authorities Branch Package*
    This additional licence offers extra functions for municipalities, communities and ministries.
  - *Mitel OpenCount Functional Upgrade to Comfort*
    This additional licence offers extra functions such as PIN telephony.
  - *Mitel OpenCount Functional Upgrade to Premium*
    This additional licence offers extra functions such as intermediate statements, invoicing etc.
  - *Mitel OpenCount Users*
    This additional licence enables a defined number of users to be monitored via

OpenCount. All OpenCount users must be licensed, otherwise a warning is generated.

Note:
Either the OpenCount application or a third party application can use the Open Application Interface.

Interfaces

- *ATAS Interface / ATASpro Interface*
  With ATAS licences external alarm and messaging sources can be connected via the Ethernet interface. The licences also offer additional possibilities compared with ATPCx
  ATAS Interface: Many commands available for messaging (displaying text and presenting softkeys on system phones), emergency number called alarm, safeguard basic with Redkey, charging bay monitoring etc.
  ATASpro Interface: Additional functions available like DECT localization, public emergency number called alarm, evacuation alarm, enhanced safeguard with alarm trigger, get rooms and room state.

  Note:
  If you use the Mitel Open Interfaces Platform, OIP takes the licences from the communication server. So always acquire these licences for the communication server so you can use ATAS even without OIP.
- *BSS Licence*
  This licence allows a BluStar server to be connected.
- *BSS-Lync Interface*
  This licence allows the use of the BluStar Lync interface.
- *CSTA Sessions*
  This licence allows third-party applications to monitor/check a terminal on the communication server using the CSTA protocol. If a terminal is monitored or checked by several applications or instances, one licence is required for each monitoring/check.
- *Presence Sync. via SIMPLE and MSRP*
  SIMPLE (Session Initiation Protocol for Instant Messaging and Presence Leveraging Extensions) is a protocol for exchanging presence information, and is used between SIP endpoints (terminals, network interfaces and nodes). MSRP (Message Session Relay Protocol) is a protocol used for exchanging data between SIP clients (e.g. for chats). This combined licence defines the number of users who can use one or both protocols for third-party applications. Only one licence is needed for a user with several SIP phones.
- *Basic User*
  This licence allows third-party applications to use the Open Application Interface.

Note:
Either the OpenCount application or a third party application can use the Open Application Interface.

3. 4. 7 Restricted operating mode

Without a valid *Software Release* licence the communication server switches over to a restricted operating mode four hours after each restart. The restriction concerns the following aspects:

Restricted operating features:

- No call information for incoming calls and during the call connection.
- Dialling by name is deactivated.
- Functions cannot be invoked via the menu or function key (likewise no enquiry calls can be made).
- The team keys do not work.
- Functions codes are not carried out (except remote maintenance on/off).
- Dialling from PC and other CTI functions are not supported.

Restricted services and routing functions:

- Calls are not routed to integrated mobile/external phones.
- Call centre functions are out of service (no routing to ACD).
- Voice mail functions are out of service (no call routing to voice mail).
- The announcement service is out of service.

3. 4. 8 Temporary offline licences

If the connection to the master is interrupted in an AIN, the satellites restart in offline mode. The licences acquired on the master are no longer visible for the satellites in offline mode. To ensure autonomous VoIP and QSIG traffic temporarily, certain licences are enabled in the satellites concerned for the duration of offline operation or for a maximum of 36 hours (the licences are not visible in WebAdmin). The licence overview (Tab. 33) shows which licences are affected. To ensure longer offline operation, the necessary licences must also be acquired on the satellites.

3. 4. 9 Trial licences

Trial licences are available for some functions. This means that functions or features that require a licence can be used and tested, licence-free, for a period of 60 days. The trial licences are automatically enabled the first time a particular function is used and then listed in WebAdmin in the *Licences* (=q9) view, complete with the date on which they expire. This procedure can only be used once for each function or feature. Thereafter you must acquire a licence. The licence overview (Tab. 33) shows which trial licences are available.

Overview of licences

Tab. 33 Overview of licences

Licence	Licensed attributes	With-out licence	With licence	Licences for net-working	Offline licence	Trial licence
Software						
Software Release	Allows a particular soft-ware release to be oper-ated	Restricted[1]	Unrestricted	In the AIN, only on the Master; otherwise per node.	–	–
Users						
User	Allows user operation on Mitel 470.	Locked	1, 20, 50, 100 or 200 additional users per licence.	In the AIN, only on the Master; otherwise per node.	✓	–
Basic User	Licence bundle: 1 additional user 1 phone licence (any one) 1 phone per user only	0	1 additional user per licence.	In the AIN, only on the Master; otherwise per node.	✓	–
Entry UCC User	Licence bundle: • 1 additional user • 8 phone licences (any one) • 8 phones per user • Video licence for all licensed phones. • MiCollab role *UCC Entry*	0	1 additional user per licence.	In the AIN, only on the Master; otherwise per node.	✓	–
Standard UCC User	Licence bundle: • 1 additional user • 8 phone licences (any one) • 8 phones per user • Video licence for all licensed phones. • MiCollab role *UCC Standard*	0	1 additional user per licence.	In the AIN, only on the Master; otherwise per node.	✓	–

Licence	Licensed attributes	With-out licence	With licence	Licences for net-working	Offline licence	Trial licence
Premium UCC User	Licence bundle: • 1 additional user • 8 phone licences (any one) • 8 phones per user • Video licence for all licensed phones. • MiCollab role *UCC Premium*	0	1 additional user per licence.	In the AIN, only on the Master; otherwise per node.	✓	–
Terminals						
MiVoice 2380 IP Softphones	Number of registered MiVoice 2380 IP IP soft-phones	0	Per licence 1 additional IP softphone	In the AIN, only on the Master; otherwise per node.	✓	✓
MiVoice 5300 IP Phones[2)]	Number of registered , MiVoice 5360 IP, MiVoice 5361 IP, MiVoice 5370 IP and MiVoice 5380 IP IP system phones	0	1, 20 or 50 additional IP system phones per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel SIP Terminals	Number of registered phones of the Mitel 6800/6900 SIP series	0	1, 20 or 50 additional Mitel SIP phone per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel 8000i Video Options	Use of the video functionality of an Mitel SIP terminal	0	Additional licence for *Mitel SIP Terminals*. 1, 20 or 50 additional Mitel SIP terminals with video functionality per licence.	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel Dialog 4200 Phones	Number of registered Dialog 4220, Dialog 4222 and Dialog 4223 digital phones	0	One additional phone per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
*MMC Extension*s	Number of mobile phones that can be registered with Mitel Mobile Client for operation with an Mitel Mobile Client Controller (MMCC)	0	Per licence 1 additional mobile phone (with Mitel Mobile Client)	In the AIN, only on the Master; otherwise per node.	–	–

Licence	Licensed attributes	With-out licence	With licence	Licences for net-working	Offline licence	Trial licence
Dual Homing	Number of registered Mitel 6800/6900 SIP phones on a backup communication server	0	Per licence 1, 20 or 50 additional phones	Always on the backup communication server	–	✓
Mobile or External Phone Extensions	Number of mobile/external phones that can be registered (without Mitel Mobile Client)	0	One additional mobile/external phone per licence (without Mitel Mobile Client)	In the AIN, only on the Master; otherwise per node.	✓	✓
SIP Terminals	Number of registered standard SIP terminals	0	1 additional standard SIP terminal per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
Video Terminals	Use of the video functionality of a standard SIP terminal	0	Additional licence for *SIP Terminals*. 1 additional standard SIP terminal with video functionality per licence.	In the AIN, only on the Master; otherwise per node.	✓	✓
BluStar						
BluStar Softphones	Number of registered BluStar softphones	0	1, 20 or 50 additional BluStar softphones per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
BluStar Softphone Video Options	Use of the video functionality of a BluStar softphone	0	Additional licence for BluStar softphone. 1, 20 or 50 additional BluStar softphones with video functionality per licence.	In the AIN, only on the Master; otherwise per node.	✓	✓
Audio services						
Conference Bridge (Dial-In Conference)	Use of conference bridge	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓
Number in Queue	Use of the function 'queue with announcement '	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓

Licence	Licensed attributes	With-out licence	With licence	Licences for net-working	Offline licence	Trial licence
Auto Attendant	Use of the auto attendant function	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Enterprise Voice Mail	Voice compression, expanded voice memory capacity, and e-mail notification whenever new voice messages are received, forwarding of voice messages, call recording.	Locked	Enabled (including 2 audio channels for voice mail, Auto Attendant or call recording)	In the AIN, only on the Master; otherwise per node.	✓	✓
Audio Record & Play Channels	Audio channels for recording or playing back audio data.	Locked	Per licence 1 additional audio channel for voice mail, Auto Attendant or call recording	In the AIN, only on the Master; otherwise per node.	–	–
Features						
Analogue Modem	Use of the modem functionality on an Mitel 415/430.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Secure VoIP	Encrypted VoIP connections using SRTP and TLS.	Non-encrypted transmission	Encrypted transmission	Per node	–	–
Silent Intrusion	Use of the Silent intrusion feature	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	–
Resources						
Mitel 470 Base licence[3)]	Allows the operation of Mitel 470 with 20 users. No additional licences needed for setting up a AIN.	Restricted[1)]	Unrestricted with up to 20 users (also in a AIN).	In the AIN, only on the Master; otherwise per node.	✓	–
VoIP Channels for Standard Media Switch[4)]	VoIP functionality	0 / 2 [5)]	Per licence 1 additional VoIP channel	In the AIN, only on the Master; otherwise per node.	✓	✓
Network						

Licence	Licensed attributes	Without licence	With licence	Licences for networking	Offline licence	Trial licence
Lync Option for SIP Access Channels	Enables using a SIP access channel with Lync-specific options and features.	0	Additional licence for *SIP Access Channels*. Per licence one additional channel with Lync-specific options and features.	In the AIN, only on the Master; otherwise per node.	✓	✓
B-Channels on PRI Cards	B channels that can be used simultaneously on the PRI interface	10	Per licence 1 additional B- channel	In the AIN, only on the Master; otherwise per node.	–	–
SIP Access Channels	Simultaneously usable channels to an SIP provider	0	Per licence 1 additional SIP access channel	In the AIN, only on the Master; otherwise per node.	✓	✓
Private networking						
QSIG Networking Channels[6)]	QSIG channels	0	Per licence 4 or n QSIG channels (n limited by the system capacity)	Per node	✓	✓
Applications						
Advanced Messaging	SMPP protocol for integration of an SMS server and registration of 9d cordless phones as system phones. (Includes licence SMPP)	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	–
CTI First Party via LAN	First-party CTI clients with basic functions on Ethernet interface	0	Enabled for a specific number of users (see "General system capacity", page 67)	In the AIN, only on the Master; otherwise per node.	–	✓
Dialers	Number of simultaneously active, user-linked Mitel Dialer applications.	0	1, 20 or 50 additional instances per licence	In the AIN, only on the Master; otherwise per node.	–	✓
CPU2 Fax Base	Send/receive fax messages and configure users with fax mailboxes.	0	2 *CPU2 Fax Channels* and 10 *CPU2 Fax Clients* licences.	In the AIN, only on the Master; otherwise per node.	–	–

Licence	Licensed attributes	Without licence	With licence	Licences for networking	Offline licence	Trial licence
CPU2 Fax Channels	Additional fax media channel.	0	Per licence 1 additional fax media channel (max. 8)	In the AIN, only on the Master; otherwise per node.	–	–
CPU2 Fax Clients	Additional users with fax mailboxes.	0	1, 20 or 50 additional fax mailboxes per licence	In the AIN, only on the Master; otherwise per node.	–	–
Hospitality Manager	Use of Mitel 400 Hospitality Manager	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓
Hospitality PMS Interface	Use of the PMS interface and therefore the FIAS protocol.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓
Hospitality PMS Rooms	Number of rooms when using the PMS interface.	0	1, 20, 50 or 100 rooms per licence	In the AIN, only on the Master; otherwise per node.	–	✓
Mitel OpenCount Basic Package	Basic licence: Prerequisite for all other OpenCount licences. Enables connection to the MiVoice Office 400 and the use of basic functions.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel OpenCount Healthcare Branch Package	Additional licence: Offers extra functions for care homes and retirement homes.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel OpenCount Public Authorities Branch Package	Additional licence: Offers extra functions for municipalities, communities and ministries.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓

Licence	Licensed attributes	With-out licence	With licence	Licences for net-working	Offline licence	Trial licence
Mitel OpenCount Functional Upgrade to Comfort	Additional licence: Offers extra functions such as PIN telephony.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel OpenCount Functional Upgrade to Premium	Additional licence: Offers extra functions such as intermediate statements, invoicing etc.	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	✓	✓
Mitel OpenCount Users	Additional licence: Enables a defined number of users to be monitored via OpenCount.	0	1, 20 or 50 additional users per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
Interfaces						
ATAS Interface	Use of the ATAS interface	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓
ATASpro Interface	Use of the ATASpro interface	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓
BSS Licence	Right to connect a BluStar server	unavailable	enabled	In the AIN, only on the Master; otherwise per node.	–	–
BSS-Lync Interface	Right to use the BluStar Lync interface	unavailable	enabled	In the AIN, only on the Master; otherwise per node.	–	–

Licence	Licensed attributes	Without licence	With licence	Licences for networking	Offline licence	Trial licence
CSTA Sessions	Number of monitored terminals via the CSTA protocol.	0	1, 20, 50 or 100 CSTA sessions per licence	In the AIN, only on the Master; otherwise per node.	✓	✓
Presence Sync. via SIMPLE and MSRP	Number of users who can use one (or both) protocols for the third-party applications.	0	1, 20 or 50 additional users per licence who may use both protocols.	In the AIN, only on the Master; otherwise per node.	✓	✓
OAI Interface	Use of the Open Application Interface	Locked	Enabled	In the AIN, only on the Master; otherwise per node.	–	✓

1) 4 hours after the new software has been uploaded or after a restart operation, the communication server switches over to a restricted operating mode (see "Restricted operating mode", page 83).

2) The licences can also be used if the *Mitel SIP Terminals* licences are missing.

3) This licence is not viewable in the licence overview in WebAdmin.

4) If Virtual Appliance is used as Master, the VoIP channels of the master node are made available without a licence from the integrated Mitel Media Server. However, for the satellites' VoIP channels, the licences must be purchased.

5) If VoIP mode is set to G.711, two G.711 VoIP channels per system can be used without a licence.

6) For Virtual Appliance this licence is only relevant to the QSIG networking of an AIN satellite.

All the licences are offered in separate licence packages. Depending on the sales channels the packages may differ from the licences in Tab. 33. The systems ship out unlicensed. Back-licensing is not provided for.However, resetting to the factory setting is possible.

OIP licences

OIP licences are managed by OIP itself. A detailed description of the OIP licences can be found in the System Manual Mitel Open Interfaces Platform.

3. 4. 10 Power supply capacity

The maximum number of terminals connected to the system can be limited by the supply power available for terminals. It is also important to take note of the maximum load per terminal interface.

3. 4. 10. 1 Supply power available for terminals

The internal power supply unit (PSU2U) is rated for the power requirements of a typical system expansion. An external auxiliary power supply unit (APS2) is used for purposes of redundancy or if a large number of terminals are operated without their own power supply. It can either handle the power supply on its own or be used in combination with the internal power supply unit (see also the overview "Powering the communication server", page 106).

Tab. 34 Available power output for various types of power supply

	Internal power supply unit only	External auxiliary power supply unit only	Internal power supply unit + external auxiliary power supply unit
Available power output (P_{total})	120 Watt	240 Watt	360 Watt

To calculate the power output available for the connected terminals ($P_{terminals}$) you need to deduct from the power specifications in Tab. 34 (P_{total}) the power consumption of the basic system, the interface cards, the DSP modules, the IP media modules, the CPU2 applications card and the redundant fan unit (P_{hw}).

Tab. 35 Power requirements of Mitel 470 hardware components

Designation	Output P [W]
Basic system with CPU1 call manager card	10
Interface card 1PRI/1PRI-T1[1)]	1.5
Interface card 2PRI	2
Interface card 4BRI	1
Interface card 8BRI	1
Interface card 4FXO	1
Interface card 8FXO	1.5
Interface card 16FXO	2.5
Interface card 4FXS	1.5
Interface card 8FXS	2
Interface card 16FXS	3
Interface card 32FXS	4.5
Interface card 8DSI	2
Interface card 16DSI	3
Interface card 32DSI	4
DSP module SM-DSPX1, SM-DSP1	0.75
DSP module SM-DSPX2, SM-DSP2	1.5
IP Media module EIP1-8	2
IP Media module EIP1-32	2.5

Designation	Output P [W]
4TAX, 8TAX, 16TAX call charge module	0.1
Applications card CPU2	21[2)]
Redundant fan unit RFU	3.5

1) 1PRI not for USA/Canada, 1PRI-T1 only for USA/Canada.

2) Up to 9 W more if the front-side USB interfaces are connected.

The basic system and the interface cards generate their own local power supply with an 80% efficiency. The calculated value must therefore be multiplied by a factor of 0.8 at the end. The calculation formula is therefore as follows:

$P_{terminals} = (P_{total} - P_{hw}) \times 0.8$

The total power requirements of all connected terminals must not exceed the value $P_{terminals}$.

The number of permissible terminals per system depends on the power requirements of the individual terminals. Tab. 36 provides details of the average power requirements of the terminals.

Note

The actually required power supply depends strongly on the call traffic, the wire diameter and the line length to the connected terminals. The values in the following table are average values under the following assumption:

- Phones traffic volume: Call Connection 38%, Ringing 2%
- SB-4+ radio unit: Active call connection on 2 channels
- SB-8 radio unit: Active call connection on 4 channels
- Background lighting MiVoice 5380: 30% active
- LEDs on terminals and expansion key modules: 20% active.
- Wire diameter: 0.5 mm
- Line length: 200 m

The table below shows the average power requirements of the terminals for a line length of approx. 200 m and a wire diameter of 0.5 mm.

Tab. 36 Average power requirements of terminals

Terminals	Socket	Output P [mW]
MiVoice 5360[1)]	DSI-AD2 interface	280
MiVoice 5361	DSI-AD2 interface	680
MiVoice 5370	DSI-AD2 interface	680
MiVoice 5380	DSI-AD2 interface	820
MiVoice 5370, MiVoice 5380 with power supply unit	DSI-AD2 interface	0
Expansion key module MiVoice M530	MiVoice 5370	110
Expansion key module MiVoice M530	MiVoice 5380	120
Expansion key module MiVoice M535	MiVoice 5370, MiVoice 5380	0 [2)]
Dialog 4220	DSI-DASL interface	390

Terminals	Socket	Output P [mW]
Dialog 4222	DSI-DASL interface	640
Dialog 4223	DSI-DASL interface	660
EKP expansion key module	Dialog 4222, Dialog 4223	45
Radio unit without power supply unit SB-4+	DSI-AD2 interface	1500 [3)]
Radio unit without power supply unit SB-8	2 DSI-AD2 interfaces	1350 [4)]
Radio unit with power supply unit SB-4+/SB-8	1 or 2 DSI-AD2 interfaces	< 100
ISDN terminal	BRI-S interface	approx. 500 [5)]
Analogue terminals	FXS interface	approx. 500

1) Although no longer available, the phone is still supported.

2) An MiVoice M535 always requires a power supply unit

3) The value applies to radio units with hardware version "-2". The value for hardware version "-1" is 300 mW lower.

4) The value applies to each interface and to radio units with hardware version "-2". The value per interface for radio units with hardware version "-1" is 150 mW lower.

5) The value depends greatly on the terminal type.

Tip
With the planning application Mitel CPQ the power supply available for terminals is checked automatically.

Overload shut-down

If 80% of the available power output is exceeded, the event message *Terminal power supply overload* is generated.

If 100% of the available power output is exceeded, the event message *Terminal power supply shut-down* is generated. The power supply is then shut down step by step, starting with the expansion slots with the highest numbers and, within the cards, with the ports with the highest numbers. The terminal ports (FXS, DSI, BRI-S) are shut down in groups of 4 ports. The exchange ports (PRI, BRI-T, FXO) are never shut down.

Once the power required drops below 100% as a result of the shut-downs, the disconnected ports are reconnected after approx. 10 seconds. If the limit of 100% is again exceeded, the overload shut-down is triggered once again.

The overload shut-down works in principle for all three types of power supply (see Tab. 34). However it triggers particularly in cases where only the internal power supply unit is available and a large number of terminals are operated without their own power supply.

If an overload occurs, either reduce the required supply power (e.g. by powering DECT radio units and or system phones locally) or use the external auxiliary power supply unit for terminals.

3. 4. 10. 2 Power supply per interface

DSI interface card

The maximal available power supply on the DSI ports per interface is limited. In certain cases (e.g. 32 connected SB-4+ radio units with HW version " - 2" at a 32DSI interface during simultaneously high traffic load) this value can be exceeded and the the overload shut-down is triggered. To provide remedy individual terminals must either be powered locally or spread out on several DSI interface cards.

Tab. 37 Maximal power supply per interface card

Maximal power supply per interface card	Output P [W]
DSI interface card	41.5

3. 4. 10. 3 Power supply per terminal interface

The power supply per terminal interface is determined by the interface type. The interface load depends on the following variables:

- Terminals used incl. auxiliary devices
- Bus configuration
- Line length and conductor cross-section

For information on the calculations refer to "Terminal interfaces", page 135.

4 Installation

This Chapter tells you how Mitel 470 can be installed and the conditions to be observed. It also includes the mounting into a 19” rack, the correct way to connect the earthing, and the power supply. Other topics in this chapter include how to fit system modules and interface cards. Finally the Chapter also describes the network- and terminal-side connection of the interfaces and the installation, powering and connection of system terminals.

4. 1 System components

The figure below shows the components of the Mitel 470 communication server complete with the additional options.

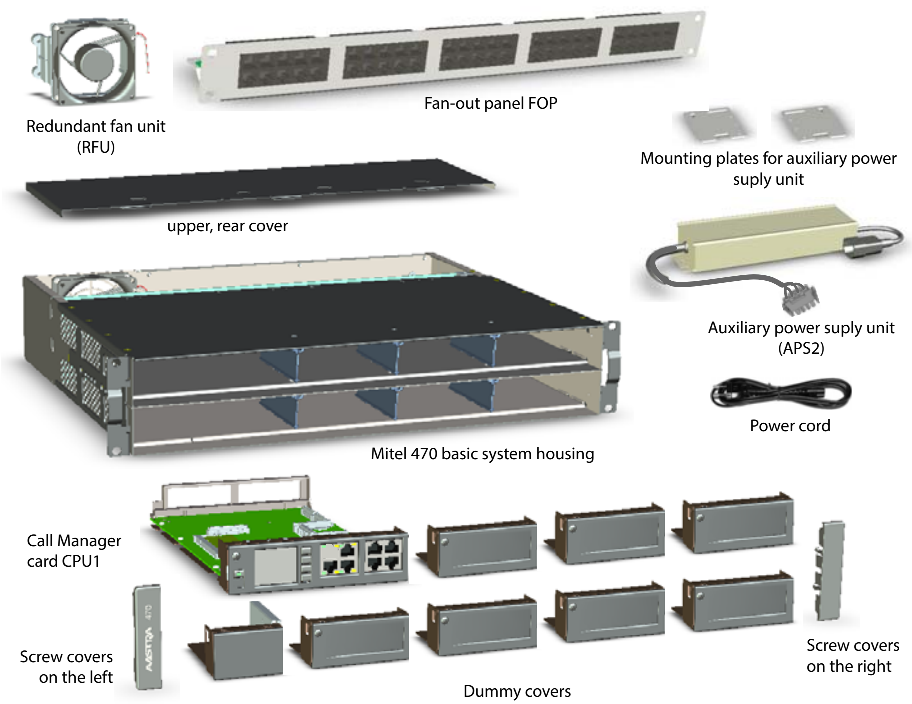


Fig. 17 System components with mounting options

4. 2 Fitting the communication server

The Mitel 470 communication server is designed for installation in a 19” rack (2 height units). The communication server can also simply be placed on a flat surface. Wall-mounting is not allowed.

4. 2. 1 Equipment supplied

The equipment supplied with the Mitel 470 includes:

- Mitel 470 communications server with integrated call manager card
- Fastening kit for rack mounting
- 2 covers for the rack screws
- 4 rubber feet for desktop installation
- Power cord
- Product information

4. 2. 2 Location requirements

The following location requirements must be observed when positioning the communication server.

WARNING!
Failure to observe the location requirements can cause the communication server to overheat, damaging electrical components and/or the surrounding area.
An event message is generated if the heat dissipation is insufficient. Appropriate measures must then be taken immediately to improve heat dissipation, e.g. providing the required clearances or lowering the ambient temperature.

Tab. 38 Mitel 470 Location requirements

Heat radiation	• Do not position in direct sunlight, near radiators or near other heating sources
EMC	• Do not position in strong electromagnetic fields of radiation (e.g.near x-ray equipment, welding equipment or similar).
Heat dissipation	• With desktop and rack mounting the ventilation holes (left) and the fan outlet (rear) must not be obstructed. • All the communication server’s housing openings must always be closed during operation to ensure a controlled flow of air (see Fig. 18).
Environment	• Ambient temperature 5 °C...45 °C • Relative humidity 30…80%, non-condensing

4. 2. 3 Safety regulations

Be sure to observe the following safety regulations before carrying out work inside a communication server:

WARNING!
Once the communication server is connected to the mains, there are hazardous voltages inside the housing. Always observe the following points before removing the housing cover:
- Disconnect the communication server from the power supply.
- Wait at least one minute so the charged capacitors have time to discharge.

CAUTION!
Components, interface cards or system modules can be damaged by electrical voltage.
Always disconnect the communication server from the power supply before removing the housing cover.

CAUTION!
Components can be damaged by electrostatic discharge when touched.
Always touch the earthed metal case of the communication server before carrying out work inside the housing. This also applies to interface cards and system modules that are no longer packed inside the ESD protective wrapping.

4. 2. 4 Flow of hot air

The Mitel 470 communications server comes with a fan already pre-installed. The housing is designed so the air flow is first guided at two levels over the processor cards and the interface cards, then passes through cutouts in the backplane, absorbs the heat from the power supply unit, and exits the housing through the fan aperture.

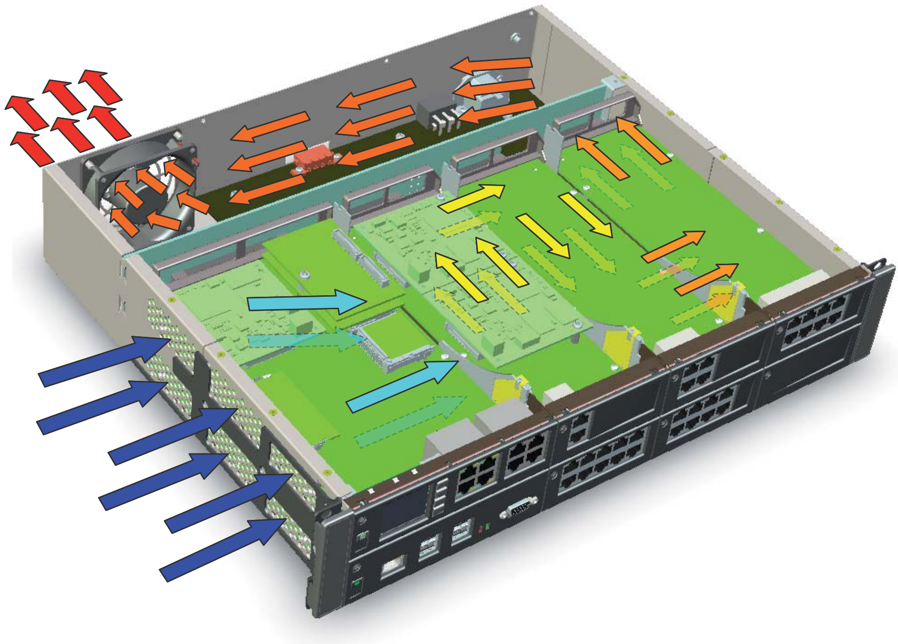


Fig. 18 Flow of hot air

The fan speed depends on the ambient temperature, the number of cards and modules, and the communication server load, and continually adapts to the current temperature inside the housing.

Note
Make sure all the housing openings on the communications server are always closed and secured by screws during operation to ensure a controlled flow of air. This applies in particular to the interface cards and processor cards, but also to the dummy covers and housing covers.

4. 2. 5 Desktop installation

For desktop installation simply place the Mitel 470 communication server on a flat, level surface. Several communication servers can be stacked directly on top of one another.

For the desktop installation of the communication server proceed as follows:

1. Affix the 4 rubber feet supplied to each of the corners of the communications server's housing base.
2. If necessary install the redundant fan unit (see "Fitting an additional fan", page 100).

3. Connect the earthing (see "Connecting the earthing wire", page 104).
4. Always observe the location requirements set out in Tab. 38.

4. 2. 6 Rack-mounting

The rack mounting of the Mitel 470 communication server allows it to be installed horizontally in a 19” rack. Be sure to observe the following:

- The communication server takes up the space of 2 height units inside the 19” rack. (1 height unit corresponds to 44.45 mm).
- Several communication servers can be stacked directly on top of one another. To do so, make sure the rubber feet are removed first.
- With interface cards with more than 8 ports it is advisable to route the cabling via an fan-out-panel (FOP) (1 height unit).

4. 2. 6. 1 Rack-mounting procedure

Materials required:

- Fastening kit for rack mounting
- Screwdriver

To rack-mount a communication server proceed as follows:

1. Pull off the screw covers on the left and right of the front panel.
2. Secure the cage nuts in the appropriate positions in the rack’s fastening rails.
3. If necessary install the redundant fan unit (see "Fitting an additional fan", page 100).
4. Connect the earthing (see "Connecting the earthing wire", page 104).
5. Secure the communications server to the rack’s fastening rails using the M6 screws and the cage nuts.
6. Fit the screw covers on the left and right of the front panel.
7. Connect the earthing (see "Connecting the earthing wire", page 104).
8. Always observe the location requirements set out in Tab. 38.

4. 2. 6. 2 Fitting an additional fan

An additional fan can be fitted in front of the standard fan already integrated. Both fans always rotate at the same time and at the same speed, depending on the temperature inside the communications server. The redundant fan unit increases the system’s operating reliability. If one fan fails, the second fan dissipates the heat. A fan failure generates an event message, allowing the defective fan (or both fans) to be replaced.

Note
Fans have a limited service life. So if a fan fails become of age (approx. 5 years) it is advisable to replace both fans as a precautionary measure.

Materials required:

- Mitel 470 additional fan pre-mounted on fastening frame
- Set of screws for additional fan
- Screwdriver

To install the additional fan proceed as follows:

1. Shut down the communication server via the control panel (see "Call-Manager display and control panel", page 218) and disconnect it from the power supply.

CAUTION!
Be sure to observe the "Safety regulations", page 98.

2. Remove the upper rear housing cover.
3. Remove the 4 rubber covers from the holes in the back panel of the communications server provided for mounting the additional fan.
4. Use the 4 enclosed screws to fit the fastening frame complete with additional fan to the back panel of the communications server (see Fig. 19).
5. Plug the fan connector into the connector marked "FAN 2" on the internal power supply unit.
6. Fit the upper rear housing cover. In so doing follow the instructions on how to ensure that the backplane BP2U sits correctly, on page 102 and the corresponding diagram (Fig. 20).
7. Reconnect the communication server to the power supply.

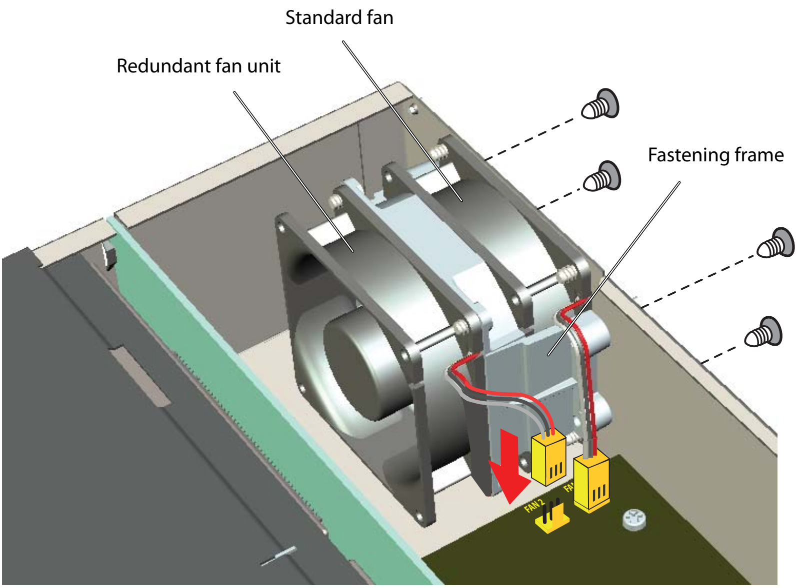


Fig. 19 Fitting the additional fan in Mitel 470

How to make backplane BP2U sit correctly

When the rear housing cover is open (e.g. so an additional fan can be installed), the backplane can spring out from the lower guide carriages (above all if no card is installed).

Result:

after the assembly, this may not allow cards to be plugged in / make real contact / be detected, etc.

Remedy:

- Check that the backplane is sitting properly in the 4 lower guide carriages. In any case, you must press down the backplane slightly since the contact springs create a certain counter-pressure behind the mounting brackets (see ① in Fig. 20).
- Check whether the backplane does not protrude from the upper part of the housing (see ② in Fig. 20).
- While closing the upper rear cover, check that the backplane is sitting correctly in the 4 upper guide carriages. It should be possible to close the cover without strain and without bending it (see ③ in Fig. 20).

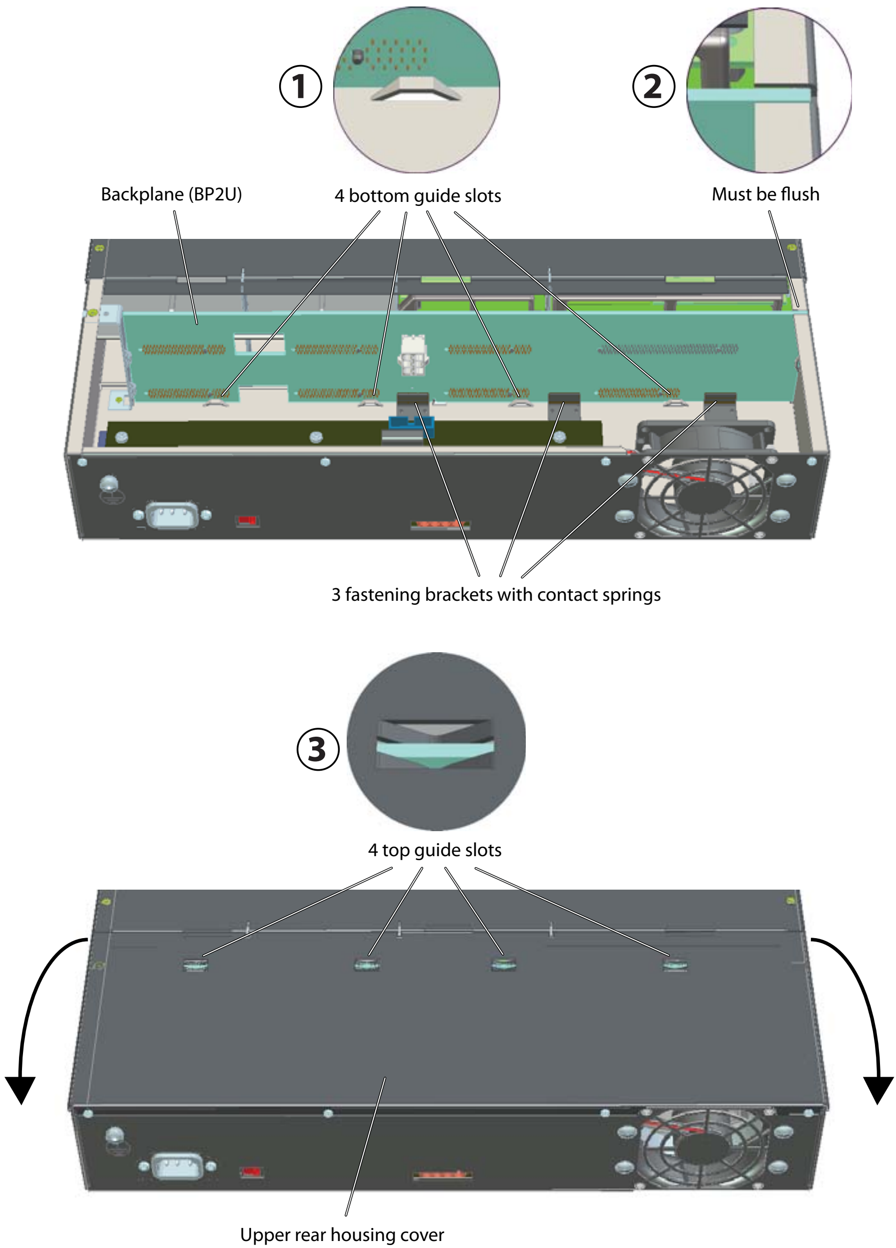


Fig. 20 Correct sitting of backplane BP2U

4. 3 Earthing and protecting the communication server

The protective earth and equipotential bonding are important integral parts of the safety concept: Standard EN 60950 relevant to safety matters stipulates protective earthing.

CAUTION!
High leakage currents can occur as a result of connecting to the communication network.
Establish an earth connection before connecting to the communications network.
Disconnect the communication server from the communications network before carrying out maintenance work.

CAUTION!
Transient overvoltage can occur on the mains and on the communications network.
Protect each line installation leading from the building by using one surge voltage protector per core at the isolating point (main) distribution frame or entry point into the building.

Operation on an IT current distribution system:
The communication server can be operated on an IT power distribution system as per EN/IEC 60950 with voltages of up to 230 VAC.

4. 3. 1 Connecting the earthing wire

The communication server's earthing connection is located on the rear panel of the communications server next to the mains power socket. The earthing wire is secured using a screw and a spring washer.

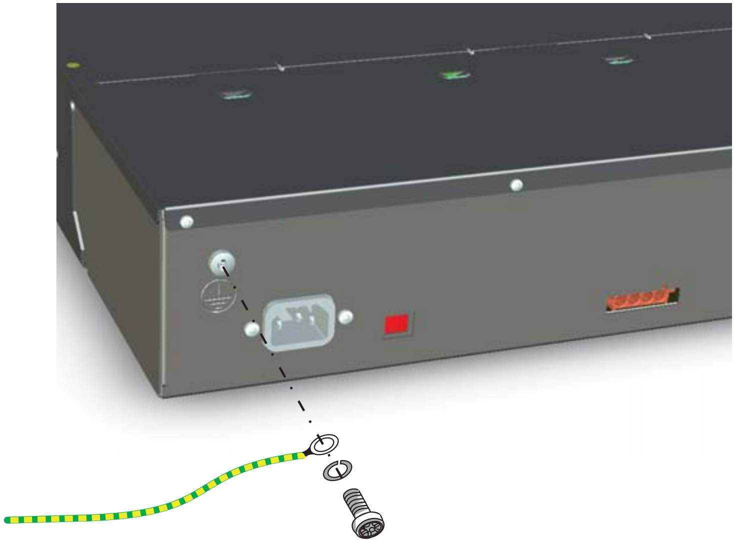


Fig. 21 Earthing connection

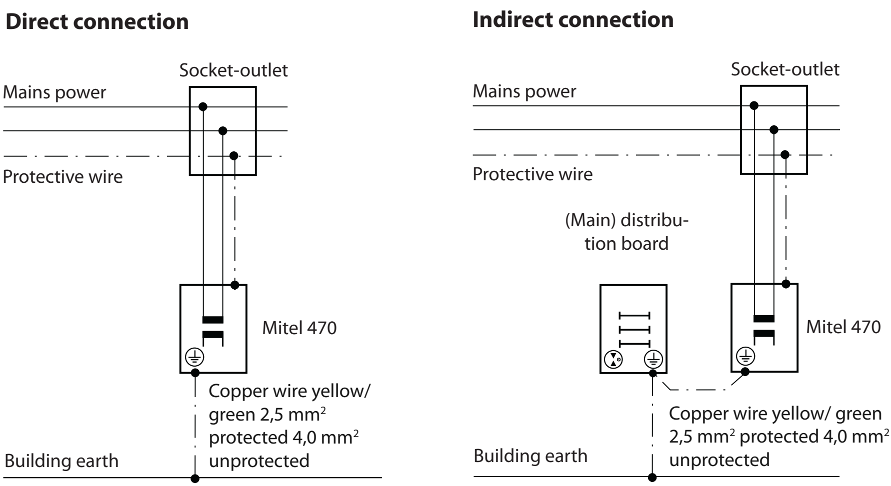


Fig. 22 Earthing of the communication server in the case of direct cabling and indirect cabling

Note
In the case of an indirection connection make sure that the communication server's earthing wire does not form any earth loops with the earthed cable screenings of the installation cables leading up to the (main) distribution frame. The cables should be kept as short as possible and laid out in parallel.

4. 3. 2 Connecting the cable screening

When using shielded installation cables also use shielded RJ45 connectors. In this way the shielding of the installation cables is automatically connected with the housing of the communication server and therefore with the building earth.

Note
Connect the cable screens to one another at the splitting point only. Observe the tree structure principle to prevent earth loops.

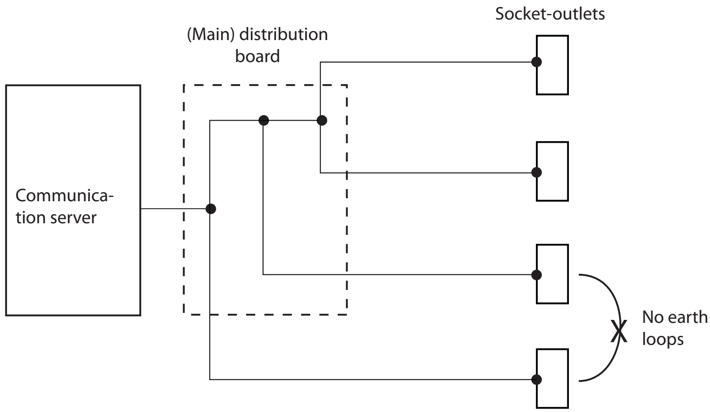


Fig. 23 Tree structure principle

4. 4 Powering the communication server

The communications server is powered as standard with 230 VAC or 115 VAC directly from the mains. The internal power supply unit (PSU2U) is rated for the power requirements of a typical system expansion. The external auxiliary power supply unit APS2 can be used to increase the power supply available or to increase operating reliability (redundancy in the event of a failure on the part of one of the two power supplies). The communication server can also be operated with the external auxiliary power supply unit only. To ensure that its operation is maintained even in the event of a mains outage, an external uninterruptible power supply (UPS) must be used.

WARNING!
Hazard due to heat generation in the event of short-circuits. The mains power supply connection must be protected with 16 A maximum in countries with 230 V mains power (for instance in Europe), and with 20 A maximum in countries with 115 V mains power (e.g. in North America).

The overview table below lists the four different types of power supply with the available power outputs:

Tab. 39 Power supply types for the communication server

Power supply type	Available power output	Redundancy operation possible	Remarks
Internal power supply unit only	120 Watt	No	Suitable for a typical system configuration
Internal power supply unit + external auxiliary power supply unit	120 Watt	yes	Suitable for a typical system configuration with power supply redundancy
External auxiliary power supply unit only	240 Watt	No	Minor heat generation inside the Mitel 470 housing
Internal power supply unit + external auxiliary power supply unit	360 Watt	No	Suitable for maximum power requirements

4. 4. 1 Internal power supply unit

The communication server is powered via the supplied mains power cord.

The following points are to be observed:

- The mains connector acts as a disconnecting device and must be positioned so that it is easily accessible.
- The voltage selector must be set to the voltage of the connected mains power (see Fig. 24).

CAUTION!
PCBs may be damaged or become defective if the communication server is operated on 230 V mains power and the voltage selector is set to 115 V or if the communication server is operated on 115 V mains power and the voltage selector is set to 230 V.

4. 4. 2 External auxiliary power supply unit

The use of the external auxiliary power supply unit APS2 is necessary to increase the operating reliability (redundancy operation) or if the internal power supply unit is no longer sufficient based on the power requirement calculations or any event messages generated (power supply overload). It is also connected directly to the 230 VAC or 115

VAC mains. However, unlike the internal power supply unit it does not have a voltage converter. The voltage automatically adapts to the mains voltage.

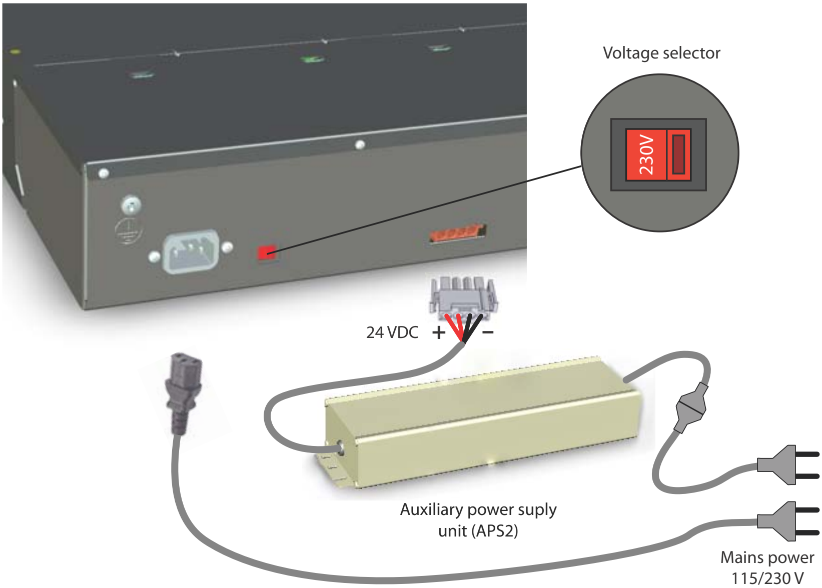


Fig. 24 Power supply to the communication server

Note

For an external power supply use the optional auxiliary power supply unit APS2 exclusively.

Tip

In redundancy operation, connect the communication server power supply and the APS2 auxiliary power supply unit to separately protected mains power. This will further enhance the system's operating reliability.

Mounting the auxiliary power supply APS2

The auxiliary power supply APS2 is supplied with a fastening kit that includes two fastening plates and 6 screws. If a fan-out-panel FOP is already fitted, the auxiliary power supply can be installed behind the connection panel.

The following diagram shows the fan-out-panel FOP from below with the auxiliary power supply fitted.

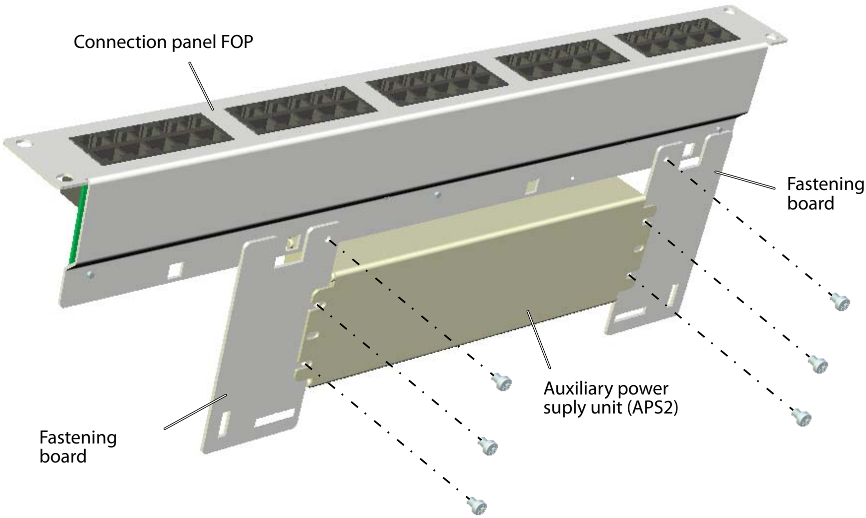


Fig. 25 Fan-out-panel with auxiliary power supply fitted (viewed from below)

4. 4. 3 Uninterruptible power supply (UPS)

The use of an external uninterruptible power supply (UPS) guarantees operation even in the event of a mains outage.

The UPS battery capacity is rated according to the communication server's primary power requirements and the required bridging time. The table below shows the maximum power requirements of the communication server in its maximum configuration and maximum traffic volume for different types of power supply.

Tab. 40 Maximum power requirements of the communication server

Communication server	Maximum power requirements
Internal power supply unit only	210 VA
External auxiliary power supply unit only	400 VA
Internal power supply unit + external auxiliary power supply unit	610 VA

The battery capacity required [Ah] can be calculated using the battery voltage and the maximum bridging time. It is important to note that the battery must never be allowed to become completely discharged and that in typical conditions only approx. 60% of the maximum power requirements is needed.

Note

The uninterrupted operation of the communication server is ensured if the UPS takes over the power supply within 20ms of the mains outage.

See also

For more technical details see "Technical data", page 260.

4. 5 Equipping the Basic System

For individual expansion the Mitel 470 basic system can be fitted with interface cards, system modules and an application card. An overview can be found in the Chapter "Expansion Stages and System Capacity", page 44.

4. 5. 1 Fitting interface cards

Interface cards are fitted to slots 2 to 8. Slot 1 is reserved for the call manager card. If an application card is fitted, slot 2 for interface cards is no longer available either.

Fig. 26 Number of the Mitel 470 slots

To fit an interface card, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Shut down the call manager via the control panel (see "On/Off key", page 218).
2. Unscrew the screw on the dummy cover and remove the cover by pulling the screw. Note: The narrow dummy cover in slot 2 is only removed when an application card is fitted.
3. Carefully slide the interface card into the slot shaft and gently press the card as far as it goes into the connection on the backplane.
4. Use the screw to secure the card in its slot.
5. Restart the call manager by pressing the On/Off button on the call manager card.

4. 5. 2 Fitting application card CPU2

The application card is wider than an interface card and can only be fitted to slot 2 (see Fig. 26).

To fit an application card, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Unscrew the screw on the larger dummy cover in slot 2 and remove the cover by pulling the screw.
2. Remove the plastic cover of the narrow dummy cover in slot 2. To do so insert a screwdriver at an angle from below to release the snap-in mechanism on the plastic cover.
3. Unscrew the screw on the narrow dummy cover and remove the cover by pulling the screw.
4. Carefully slide the application card into the shaft of slot 2 and gently press the card as far as it goes into the connection on the backplane.
5. Use the screw to secure the card in its slot.
6. Connect the cables of any assigned interfaces on the front panel of the applications card.
7. Start up the applications server by pressing the On/Off button on the applications card.

See also:
For more information about installing, configuring and upgrading the software of the application card, see the CPU2-S application card installation manual.

4. 5. 3 Equipping the call manager card CPU1

The call manager card is part of any communications server and is required for a fully functional system. It is already fitted ex-works and only needs to be removed in the event of repairs (see "Operation and Maintenance", from page 197) or when expanding the system with modules. The call manager card only fits into slot 1 (see Fig. 26).

4. 5. 4 Fitting system modules

With system modules a distinction is made between modules expandable as an option (DSP modules, IP media modules, Call charge modules) and mandatory modules (RAM module). The system cards (Flash card, EIM card) are always required.

This chapter only describes the procedure for fitting system modules that are expandable as an option (DSP module, IP Media module, call charges module). The RAM module only needs to be replaced in the event of repairs or maintenance work (see "Operation and Maintenance", from page 197).

4. 5. 5 Fitting DSP modules

DSP modules are fitted to the call manager card. A maximum of 2 DSP modules can be stacked.

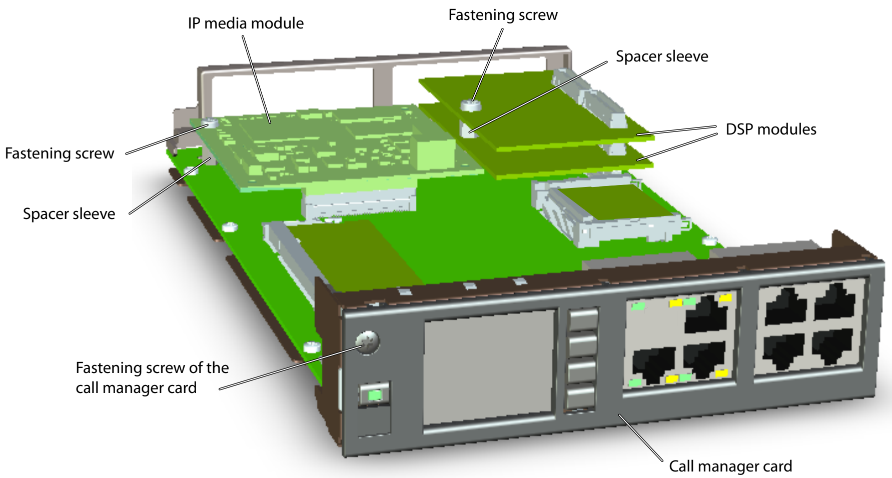


Fig. 27 **System modules on the call manager card**

To fit a DSP module, proceed as follows:

⚠ **CAUTION!**
Be sure to observe the "Safety regulations", page 98.

1. Shut down the call manager via the control panel (see "On/Off key", page 218).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the fastening screw on the module slot for DSP modules.
4. The spacer sleeve for the lower module is already pre-mounted on the processor card. For the upper DSP module screw the spacer sleeve supplied with the module into place.
5. Place the module on slot (or onto a module already fitted in that slot) and press down evenly on both connectors as far as the stop.
6. Secure the module with the fastening screw.

7. Carefully slide the call manager card into the shaft of slot 1 and gently press the card as far as it goes into the connection on the backplane.
8. Secure the call manager card back into its slot with the screw.
9. Restart the call manager by pressing the On/Off button on the call manager card.

4. 5. 6 Fitting IP Media modules

IP Media modules are fitted either to the call manager card or to PRI trunk cards. IP Media modules are **not** stackable.

To fit an IP Media module to a call manager card, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Shut down the call manager via the control panel (see "On/Off key", page 218).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the 2 fastening screws on the 2 pre-mounted spacer sleeves on the IP Media module.
4. Place the module in the slot and press it down evenly into the slot as far as the stop.
5. Fit the module on to the call manager card from below using the 2 fastening screws.
6. Carefully slide the call manager card into the shaft of slot 1 and gently press the card as far as it goes into the connection on the backplane.
7. Secure the call manager card back into its slot with the screw.
8. Restart the call manager by pressing the On/Off button on the call manager card.

Proceed accordingly to fit one or two IP Media modules to a PRI trunk card.

4. 5. 7 Fitting call charge modules

Call charge modules are fitted to FXO trunk cards. Only 1 call charge module can be fitted to each FXO card.

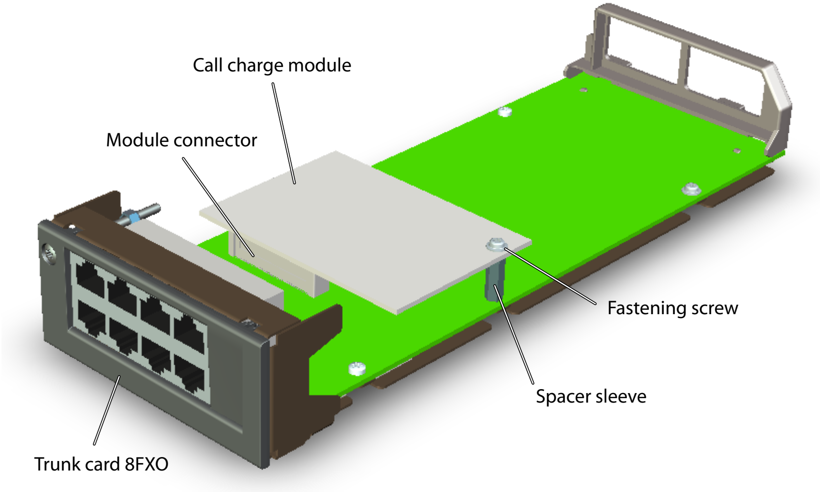


Fig. 28 Call charge module on 8FXO trunk card

To fit a call charge module, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Shut down the call manager via the control panel (see "On/Off key", page 218).
2. Unscrew the screw on the FXO card and remove the card by pulling the fastening screw.
3. Remove the fastening screw for the call charge module on the FXO card and in its place screw the spacer sleeve into position (see Fig. 28).
4. Place the module in the slot and press it down evenly into the slot as far as the stop.
5. Secure the module with the fastening screw on the spacer sleeve.
6. Carefully slide the FXO card into the slot shaft and gently press the card as far as it goes into the connection on the backplane.
7. Use the screw to secure the FXO card back into its slot.
8. Restart the call manager by pressing the On/Off button on the call manager card.

4. 5. 8 Component mounting rules

The component mounting rules mentioned in the previous chapters are listed here in an overview:

- The call manager card can only be fitted to slot 1.
- The application card can only be fitted to slot 2.
- Interface cards can be fitted to card slots 2 to 8.
  Exception: If an application card is fitted, slot 2 is no longer available for interface cards.
  Tip: Leave slot 2 empty so that it can later be equipped with an applications card, if required. This will save you a good deal of configuration work later on.
- For optimum heat dissipation interface cards should always be fitted to the basic system in the same sequence as the slot numbering (from left to right, see Fig. 26). The empty slots are therefore always those with the highest numbers (with the exception possibly of slot 2).
- Two DSP modules can be stacked and are always fitted to the call manager card.
- IP Media modules are fitted to the call manager card or to PRI trunk cards and cannot be stacked.
- The interfaces are enabled sequentially when the communication server is started up. The following rules apply:
  – The number of interfaces actually enabled is determined in each case by the system capacity (see "System capacity", page 67). If a limit value is reached, all the interface cards or all the interfaces of the last card may not be enabled.
  – The interfaces are enabled in accordance with their designation, starting with the lower designations. This means that the terminal interfaces on the processor card are always enabled before those on the interface cards.

4. 6 Connecting the communication server

There are two possibilities for connection to the telephone network and the terminal-side cabling:

- Direct connection
- Indirect cabling via (main) distribution frame and any universal building cable installation (UBC) (see also Fig. 32 and Fig. 33).

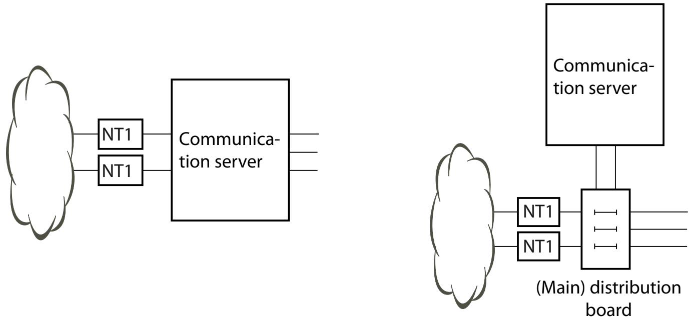


Fig. 29 Direct cabling (left) and indirect cabling (right)

On the front panel all the connections are made using RJ45 connectors.

4. 6. 1 Direct connection

Standard commercial cables are used to connect directly to the telephone network. Details can be found in the Chapter "Network interfaces", page 125.

On terminal cards with 16 or more interfaces some or all of the RJ45 sockets are multiply assigned. They can be split into individual RJ45 sockets using patch cables and the fan-out-panel (see "Fan-out panel FOP", page 157).

4. 6. 2 Indirect connection

There are two possibilities for connecting the communication server indirectly to the telephone network and terminal-side cabling:

- Connection via main distribution board
- Connection to a universal building cable installation (UBC)

4. 6. 2. 1 Connection via main distribution board

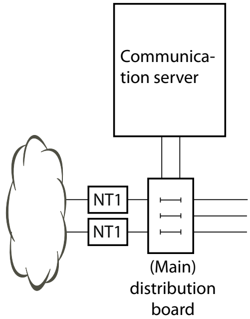


Fig. 30 Connection via main distribution board

The interface sockets on the front panel and on the fan-out-panel (FOP) where applicable are connected with the (main) distribution frame or the patch panels using either patch cables or prefabricated system cables (see "Equipment Overview", page 258).

Prefabricated system cable 4 x RJ45[1]

With terminal cards with 16 or more interfaces some or all of the RJ45 sockets are assigned four-fold on the front panel of the Mitel 470. With this cable they can be connected without the use of a fan-out-panel (FOP). The cable is 6 m long and at one extremity has four RJ45 connectors on which all the pins are wired.

Tab. 41 Schematic diagram of prefabricated system cable 4 × RJ45 × 8 Pin

Stranded element	**Core colour**	**Cable designation**	**RJ45**	**Port**
			Pin	**Two-wire connection**
1	white	1	4	x.1a
	blue		5	x.1b
	turquoise		3	x.2a
	violet		6	x.2b
2	white		1	x.3a
	orange		2	x.3b
	turquoise		7	x.4a
	violet		8	x.4b

1)Not valid for USA/Canada.

Stranded element	Core colour	Cable designation	RJ45	Port
			Pin	Two-wire connection
3	white	2	4	x.1a
	green		5	x.1b
	turquoise		3	x.2a
	violet		6	x.2b
4	white		1	x.3a
	brown		2	x.3b
	turquoise		7	x.4a
	violet		8	x.4b
5	white	3	4	x.1a
	grey		5	x.1b
	turquoise		3	x.2a
	violet		6	x.2b
6	red		1	x.3a
	blue		2	x.3b
	turquoise		7	x.4a
	violet		8	x.4b
7	red	4	4	x.1a
	orange		5	x.1b
	turquoise		3	x.2a
	violet		6	x.2b
8	red		1	x.3a
	green		2	x.3b
	turquoise		7	x.4a
	violet		8	x.4b

Prefabricated system cable 12 x RJ45[1)]

The cable is 6 m long and, at one extremity, has 12 RJ45 connectors for the interfaces on the front panel. Two of them have 4 cores; the others, 2 cores. This means the cable is suitable for connecting the following interfaces:

- 2 network interfaces BRI-T or 2 terminal interfaces BRI-S or a combination thereof.
- 10 terminal interfaces (DSI, FXS) or a combination thereof.

Note:
This cable cannot be used to connect PRI and Ethernet interfaces (see also "Connection of PRI primary rate interface", page 129 and "Connection of Ethernet interfaces", page 160).

1)Not valid for USA/Canada.

Tip
Use standard commercial connecting cables not just for the PRI and Ethernet interfaces but also for connecting the BRI-T interfaces.

Tab. 42 Schematic diagram of prefabricated system cable 12 × RJ45

Stranded element	Core colour	Cable designation	RJ45	Signal	
			Pin	Connection four-wire	Two-wire connection
1	white	1	4	f	a
	blue		5	e	b
	turquoise		6	d	–
	violet		3	c	–
2	white	2	4	f	a
	orange		5	e	b
	turquoise		6	d	–
	violet		3	c	–
3	white	3	4	–	a
	green		5	–	b
	turquoise	4	4	–	a
	violet		5	–	b
4	white	5	4	–	a
	brown		5	–	b
	turquoise	6.	4	–	a
	violet		5	–	b
5	white	7	4	–	a
	grey		5	–	b
	turquoise	8	4	–	a
	violet		5	–	b
6	red	9.	4	–	a
	blue		5	–	b
	turquoise	10	4	–	a
	violet		5	–	b
7	red	11	4	–	a
	orange		5	–	b
	turquoise	12	4	–	a
	violet		5	–	b

Prefabricated system cable 8 x RJ45 x 2 Pin[1)]

With terminal cards with 16 or less interfaces some or all of the RJ45 sockets are single assigned on the front panel of the Mitel 470. With this cable they can be connected to the main distribution board. The cable is 25 ft long and at one extremity has eight RJ45 connectors on which only 2 pins are wired.

1)Only valid for USA/Canada.

Tab. 43 Schematic diagram of prefabricated system cable 8× RJ45 × 2Pin (for USA/Canada only)

RJ45 Connector No.	Standard Pair No.	RJ45 Pin	Colour	2-wire Connection
1	1	4	white/blue	tip +
		5	blue/white	ring –
2	2	4	white/orange	tip +
		5	orange/white	ring –
3	3	4	white/green	tip +
		5	green/white	ring –
4	4	4	white/brown	tip +
		5	brown/white	ring –
5	5	4	white/slate	tip +
		5	slate/white	ring –
6	6	4	red/blue	tip +
		5	blue/red	ring –
7	7	4	red/orange	tip +
		5	orange/red	ring –
8	8	4	red/green	tip +
		5	green/red	ring –

- Examples of use for 16FXS card:
  One cable is required for ports 1...8
  Hint: Use a prefabricated system cable (4 x RJ45 x 8 Pin) to connect ports 9...16
- Examples of use for 8FXS or 8FXO card:
  One cable is required for ports 1...8
- Examples of use for 4FXS or 4FXO card:
  Half a cable is required for ports 1...4
  Hint: The remaining RJ45 connectors can be used either for another 4FXS, 4FXO or for the 4FXS ports on CPU1

Prefabricated system cable 4 x RJ45 x 8 Pin[1)]

With terminal cards with 16 or more interfaces some or all of the RJ45 sockets are assigned four-fold on the front panel of the Mitel 470. With this cable they can be connected without the use of a fan-out-panel (FOP). The cable is 25 ft long and at one extremity has four RJ45 connectors on which all the pins are wired.

1)Only valid for USA/Canada.

Tab. 44 Schematic diagram of prefabricated system cable 4× RJ45 × 8 Pin (for USA/Canada only)

RJ45 Connector No.	Standard Pair No.	RJ45 Pin	Colour	2-wire Connection
1	1	4	white/blue	tip +
		5	blue/white	ring –
	2	3	white/orange	tip +
		6	orange/white	ring –
	3	1	white/green	tip +
		2	green/white	ring –
	4	7	white/brown	tip +
		8	brown/white	ring –
2	5	4	white/slate	tip +
		5	slate/white	ring –
	6	3	red/blue	tip +
		6	blue/red	ring –
	7	1	red/orange	tip +
		2	orange/red	ring –
	8	7	red/green	tip +
		8	green/red	ring –
3	9	4	red/brown	tip +
		5	brown/red	ring –
	10	3	red/slate	tip +
		6	slate/red	ring –
	11	1	black/blue	tip +
		2	blue/black	ring –
	12	7	black/orange	tip +
		8	orange/black	ring –
4	13	4	black/green	tip +
		5	green/black	ring –
	14	3	black/brown	tip +
		6	brown/black	ring –
	15	1	black/slate	tip +
		2	slate/black	ring –
	16	7	yellow/blue	tip +
		8	blue/yellow	ring –

- Examples of use for 16FXS card:
  Half a cable is required for ports 9...16:
  - RJ45 Connector No 1 covers ports 9-12
  - RJ45 Connector No 2 covers ports 13-16
  - RJ45 Connectors No 3 and 4 are available for a second 16FXS.
    Hint: Use a prefabricated system cable (8 x RJ45 x 2 Pin) to connect ports 1...8
- Examples of use for 32FXS card (2 cables are required):

– RJ45 Connector No 1 covers ports 1-4 or ports 17-20 of a 32FXS card
– RJ45 Connector No 2 covers ports 5-8 or ports 21-24 of a 32FXS card
– RJ45 Connector No 3 covers ports 9-12 or ports 25-28 of a 32FXS card
– RJ45 Connector No 4covers ports 13-16 or ports 29-32 of a 32FXS card

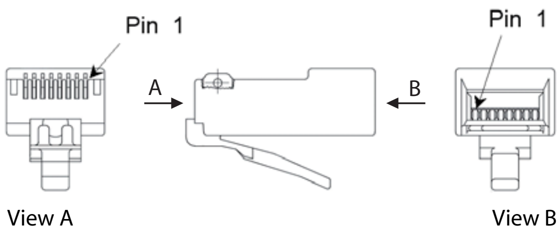


Fig. 31 Pin numbering, RJ45 connector

4. 6. 2. 2 Connection to a universal building cable installation (UBC)

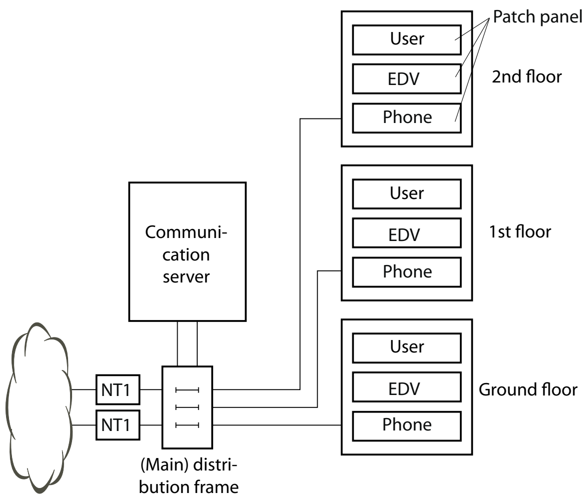


Fig. 32 Connecting to a UBC via a (main) distribution board (example)

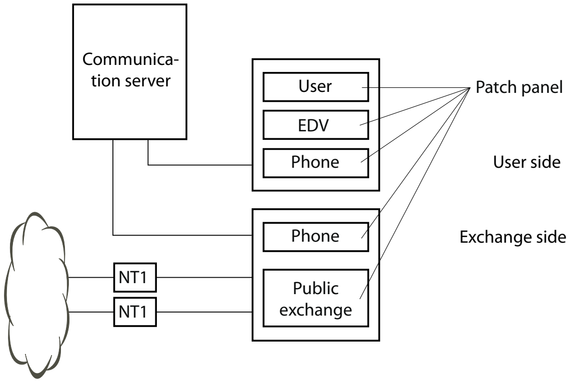


Fig. 33 Connecting to a UBC via wiring centre (example)

4. 7 Cabling interfaces

All the interfaces are routed to the front panel and are therefore accessible without opening the communication server.

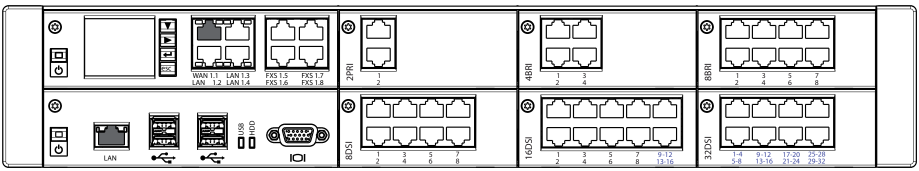


Fig. 34 Interfaces on the front panel with port designation (example)

4. 7. 1 Port addressing

A port address is always of the type x.y. x is the number of the card slot, and y, the port number.

The slot numbering starts with 1 and ends with 8 (see "Number of the Mitel 470 slots", page 110).

With BRI-S interface and DSI interface addresses, the terminal selection digit (TSD) is relevant, in addition to the slot and port numbers. This is always -1 in analogue terminal interfaces.

Tab. 45 Examples of interface addressing

Slot	Port address
Call manager card; FXS interface x.5	1.5
Interface card on slot 4; interface x.3	4.3
Terminal with TSD 2 on interface card in slot 6; interface x.4	6.4-2

4. 7. 2 Network interfaces

Equipping the system with interface cards provides the necessary network interfaces. With the exception of the Ethernet interface, which also represents a network interface via SIP access, there are no network interfaces on the Mitel 470 communication server.

4. 7. 2. 1 Basic rate interface BRI-T

Fitting BRI interface cards means that BRI network interfaces are available on the RJ45-sockets on the front panel of the cards. The possible RJ45 sockets are high-lighted in colour in the figure below.

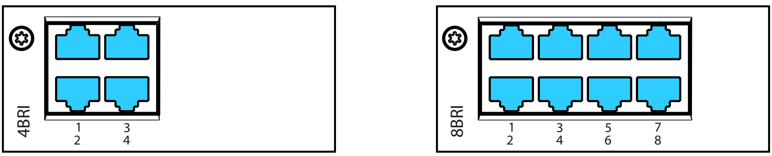


Fig. 35 Connection possibilities for BRI network interfaces

Notes

- The interfaces of sockets 1 to 4 can be switched to BRI-S. The interfaces of sockets 5 to 8 are permanently configured to BRI-T.
- Circuit type as per EN/IEC 60950: SELV
- Not usable in USA/Canada for the public network

The connection from the front panel to the NT1 (Network Termination) is via standard commercial straight patch cables with 8-pin RJ45 connectors on both sides. With the appropriate tools you can also create your own cables.

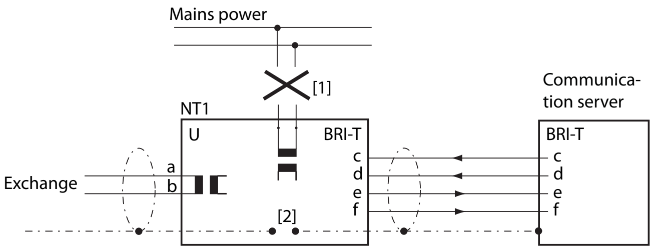
Cable Requirements

Tab. 46 Cable requirements for basic rate interface BRI-T

Core pairs × cores	1 × 4 o 2 × 2
Stranded	yes
Wire diameter, core	0.4…0.6 mm
Screening	recommended

Characteristic impedance	< 125 Ω (100 kHz), < 115 Ω (1 MHz)
Wave attenuation	< 6 dB/km (100 kHz), < 26 dB/km (1 MHz)
Near/crosstalk attenuation	> 54 dB/100 m (1 kHz to 1 MHz)

BRI basic rate interface network-side

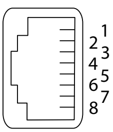
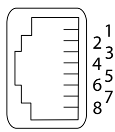


[1] Do not connect power supply NT1
[2] Do not fit the jumper

Fig. 36 Basic access on NT1

The assignment of the RJ45 connector is identical on the NT-side and on the side of the communication server.

Tab. 47 Wiring of the BRI basic rate interface network-side

NT1			Cable cores Straight patch cable	Communication server		
Socket	**Pin**	**BRI-T signal**		**BRI-T signal**	**Pin**	**Socket**
	1	–		–	1	
	2	–		–	2	
	3	c	←	c	3	
	4	f	→	f	4	
	5	e	→	e	5	
	6	d	←	d	6	
	7	–		–	7	
	8	–		–	8	

Basic access in the private leased-line network

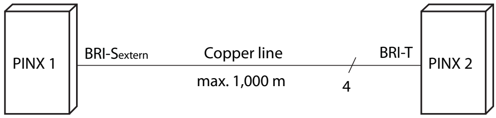


Fig. 37 BRI-S basic rate interface external, networked with copper line

Tab. 48 Connection of BRI-S basic rate interface external, networked with copper line

PINX 1 signal Basic access BRI-S ext.	Cable cores	PINX 2 signal Basic rate interface BRI-T
c	←	c
f	→	f
e	→	e
d	←	d

Bus configuration

BRI-S ext. is subject to the conditions that apply to terminal interface BRI-S (see "BRI-S terminal interfaces", page 144).

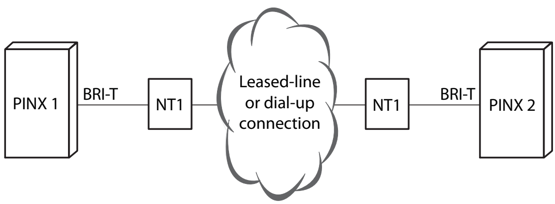


Fig. 38 Basic rate interface BRI-T, networked with leased-line or dial-up connection

Tab. 49 Cabling for basic rate interface BRI-T, networked with leased-line or dial-up connection

PINX1 signal, basic rate interface BRI-T	Cable cores	NT1	Network	NT1	Cable cores	PINX 2 signal, basic rate interface BRI-T
c	→	c		c	←	c
f	←	f		f	→	f
e	←	e		e	→	e
d	→	d		d	←	d

See also

Chapter "Connections with basic accesses" in the PISN/QSIG Networking System Manual.

4. 7. 2. 2 Primary rate interface PRI

Fitting the corresponding interface cards means that PRI network interfaces are available on the RJ45-sockets on the front panel of the cards. The possible RJ45 sockets are highlighted in colour in the figure below.

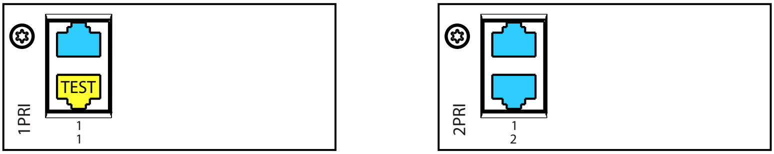


Fig. 39 Connection possibilities for PRI network interfaces

With card 1PRI/1PRI-T1[1] the PRI interface is routed in parallel to both RJ45 sockets for test purposes.

Notes

- In normal operation both sockets must not be connected on the 1PRI/1PRI-T1 card; otherwise faults may occur.
- Circuit type as per EN/IEC 60950: SELV

Cable Requirements

The connection to NT1 (Network Termination) is implemented using commercially available screened cables with 8-pin RJ45 connectors at both ends, e.g. S-FTP 4P, PVC, Cat. 5e.

Tab. 50 Cable requirements for the primary rate interface)

Core pairs × cores	2 × 2 (short distances also 1 × 4)
Stranded	yes
Wire diameter, core	0.4...0.6 mm
Screening	yes
Characteristic impedance	90 to 130 Ω (1 MHz)
Wave attenuation	< 6 dB/km (100 kHz), < 26 dB/km (1 MHz)
Near/crosstalk attenuation	> 54 dB/100 m (1 kHz to 1 MHz)

1) 1PRI not for USA/Canada, 1PRI-T1 only for USA/Canada.

PRI primary rate interface, network-side

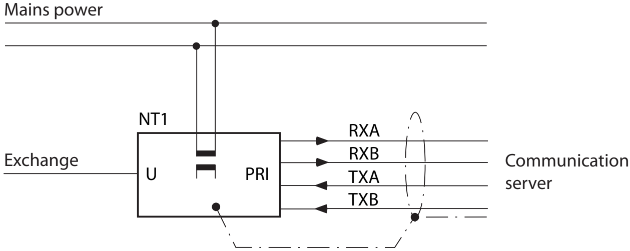
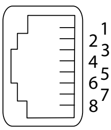
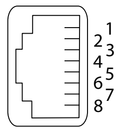


Fig. 40 PRI primary rate interface on NT1

Tab. 51 Connection of PRI primary rate interface

NT1			Cable cores Straight patch cable	Communication server		
Socket	**Pin**	**PRI signal[1)]**		**PRI signal**	**Pin**	**Socket**
	1	TxA	⟶	RxA	1	
	2	TxB	⟶	RxB	2	
	3	–		–	3	
	4	RxA	⟵	TxA	4	
	5	RxB	⟵	TxB	5	
	6	–		–	6	
	7	–		–	7	
	8	–		–	8	

1) Other designations are also possible on the NT1 such as: "S2m ab" instead of "TxA/TxB" and "S2m an" instead of "RxA/RxB".

Primary rate access in the private leased-line network

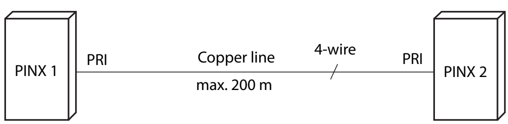
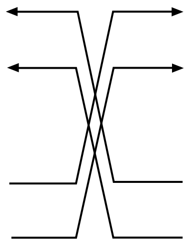


Fig. 41 Primary rate access, networked with copper line

Tab. 52 Cabling for primary rate access PRI, networked with copper line

RJ45 Pin	PRI PINX 1 signal	Cable cores Crossed patch cable	PRI PINX 2 signal	RJ45 Pin
1	RxA		RxA	1
2	RxB		RxB	2
3	—		—	3
4	TxA		TxA	4
5	TxB		TxB	5
6	—		—	6
7	—		—	7
8	—		—	8

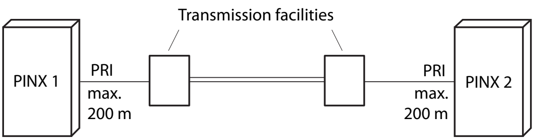


Fig. 42 Primary rate interface, networked with transmission equipment

Tab. 53 Cabling for primary rate access PRI, networked with transmission equipment

RJ45 Pin	PRI PINX 1 signal	Cable cores, straight patch cable	Transmis-sion equip-ment signal		Transmis-sion equip-ment signal	Cable cores Straight patch cable	PRI PINX 2 signal	RJ45 Pin
1	RxA	←	RxA		RxA	→	RxA	1
2	RxB	←	RxB		RxB	→	RxB	2
3	—						—	3
4	TxA	→	TxA		TxA	←	TxA	4
5	TxB	→	TxB		TxB	←	TxB	5
6	—						—	6
7	—						—	7
8	—						—	8

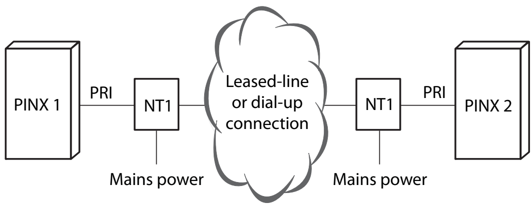


Fig. 43 Primary rate access PRI, networked with leased-line or dial-up connection

Tab. 54 Cabling for primary rate interface, PRI, networked with leased-line or dial-up connection

RJ45 Pin	PRI PINX 1 signal	Cable cores, straight patch cable	PRI signal NT1	Net-work	PRI signal NT1	Cable cores Straight patch cable	PRI PINX 2 signal	RJ45 Pin
1	RxA	←	RxA		RxA	→	RxA	1
2	RxB	←	RxB		RxB	→	RxB	2
3	—						—	3
4	TxA	→	TxA		TxA	←	TxA	4
5	TxB	→	TxB		TxB	←	TxB	5
6	—						—	6
7	—						—	7
8	—						—	8

See also:
System Manual “PISN / QSIG Networking”

4. 7. 2. 3 FXO network interfaces

Fitting the corresponding interface cards means that FXO network interfaces are available on the RJ45-sockets on the front panel of the cards. The possible RJ45 sockets are highlighted in colour in the figure below.

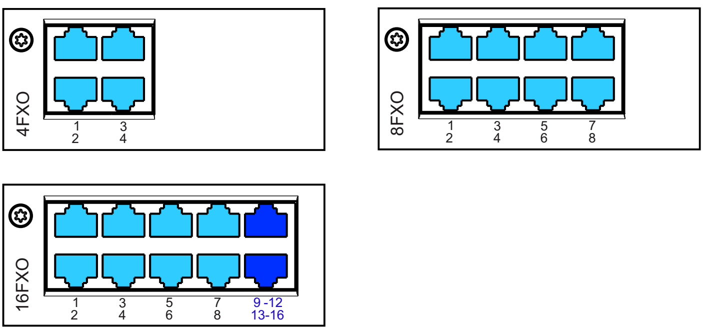


Fig. 44 Connection possibilities for FXO network interfaces

On cards with 16 interfaces RJ45 sockets 9 to 16 are multiply assigned. The signals can be split again to individual RJ45 sockets using patch cables and the fan-out panel

FOP (see "Fan-out panel FOP", page 157) or with 8-fold assigned connecting cables (see e.g. "Prefabricated system cable 4 x RJ45", page 117).

Multiply assigned RJ45 sockets are colour-coded in blue.

One call charge module can be fitted to each FXO card if required (see "Fitting call charge modules", page 113).

In a direct connection the RJ45 connector is connected directly to the trunk cable using a crimp clip.

With an indirection connection you need to observe the cable requirements.

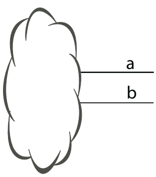
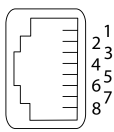
Notes

- Inadmissibly high temperatures can occur on the FXO card when connecting to local exchanges generating a very high loop current (up to 90mA). If so, the PCB temperature monitoring deactivates the FXO ports in groups of 4 ports. If the temperature then drops, the FXO ports are automatically reactivated group by group. This behaviour can occur particularly when the ambient temperature is higher than normal and/or with a system with maximum configuration. Normally local exchanges produce a loop current of approx. 25 mA, which does not cause any restrictions.
- Circuit type as per EN/IEC 60950: TNV-3

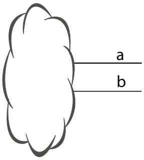
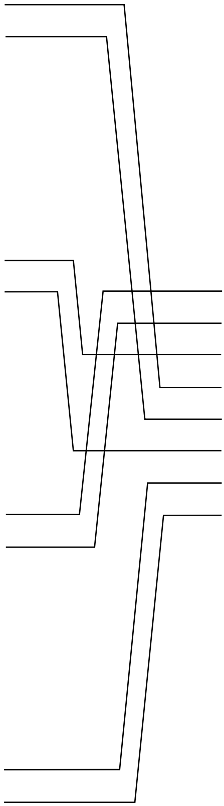
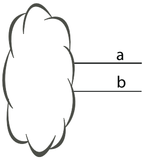
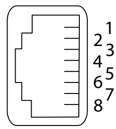
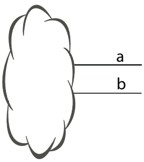
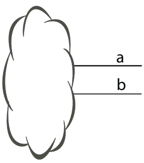
Connection

Assignment of the RJ45 sockets on the front panel:

Tab. 55 Connection FXO network interface

Public analogue network	Communication server		
	FXO signal	Pin	Socket
	–	1	
	–	2	
	–	3	
	a	4	
	b	5	
	–	6	
	–	7	
	–	8	

Tab. 56 Connection of four-fold assigned FXO network interface

Public analogue network		Splitting with fan-out panel FOP or 8-fold assigned connecting cables	Communication server		
	FXO signal		FXO signal	Pin	Socket
	–				
	–				
	–				
	1a				
	1b				
	–				
	–				
	–				
	–				
	–				
	–				
	2a				
	2b		3a	1	
	–		3b	2	
	–		2a	3	
	–		1a	4	
	–		1b	5	
	–		2b	6	
	–		4a	7	
	3a		4b	8	
	3b				
	–				
	–				
	–				
	–				
	–				
	–				
	4a				
	4b				
	–				
	–				
	–				

Cable Requirements

Tab. 57 Cable requirements for FXO network interface

Core pairs × cores	1 × 2
Stranded	not required
Wire diameter, core	0.4 … 0.8 mm
Screening	not required
Resistance	max. 2 × 250 Ω

4. 7. 3 Terminal interfaces

4. 7. 3. 1 DSI terminal interfaces

Fitting the corresponding interface cards means that DSI terminal interfaces are available on the RJ45-sockets on the front panel of the cards. The possible RJ45 sockets are highlighted in colour in the figure below.

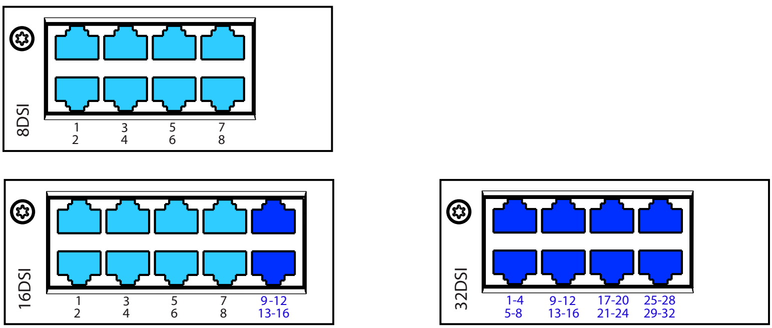


Fig. 45 Connection possibilities for DSI terminal interfaces

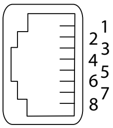
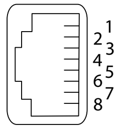
On terminal cards with 16 or more interfaces some or all of the RJ45 sockets are multiply assigned. The signals can be split again to individual RJ45 sockets using patch cables and the fan-out panel FOP (see "Fan-out panel FOP", page 157) or with 8-fold assigned connecting cables (see e.g. "Prefabricated system cable 4 x RJ45", page 117).

Tip
Multiply assigned RJ45 sockets are colour-coded in blue.

Note
Circuit type as per EN/IEC 60950: SELV

Connection

Tab. 58 Connection of individually assigned DSI terminal interface

Communication server			Cable cores	Connection socket		
Socket	**Pin**	**DSI signal**		**DSI signal**	**Pin**	**Socket**
	1	–		–	1	
	2	–		–	2	
	3	–		–	3	
	4	a		a	4	
	5	b		b	5	
	6	–		–	6	
	7	–		–	7	
	8	–		–	8	

Tab. 59 Connection of four-fold assigned DSI terminal interface

Communication server			Splitting with fan-out panel FOP or 8-fold assigned connecting cables	Connection socket		
Socket	Pin	DSI signal		DSI signal	Pin	Socket
				–	1	
				–	2	
				–	3	
				1a	4	
				1b	5	
				–	6	
				–	7	
				–	8	
				–	1	
				–	2	
				–	3	
				2a	4	
	1	3a		2b	5	
	2	3b		–	6	
	3	2a		–	7	
	4	1a		–	8	
	5	1b		–	1	
	6	2b		–	2	
	7	4a		–	3	
	8	4b		3a	4	
				3b	5	
				–	6	
				–	7	
				–	8	
				–	1	
				–	2	
				–	3	
				4a	4	
				4b	5	
				–	6	
				–	7	
				–	8	

DSI bus configuration

For each DSI interface card, in the Cards and Modules view, (*=4g*) the protocol can be chosen on the DSI bus:

- *DSI-AD2*:
  For system phones of the MiVoice 5300 series and for SB-4+ and SB-8 DECT radio units.
- *DSI-DASL*: For Dialog 4200 series system phones.

Depending on the line length 1 or 2 phones can be connected on each DSI-AD2 interface. The following requirements apply with regard to the bus length to ensure that the maximum permissible signal delay is not exceeded:

Tab. 60 DSI-AD2 bus length and number of phones

Number of phones	Total length of DSI-AD2 bus	Distance between the 1st and 2nd connection point (without connection cable)
1	A: max. 1200 m	–
2	B: max. 1200 m	C: max. 10 m

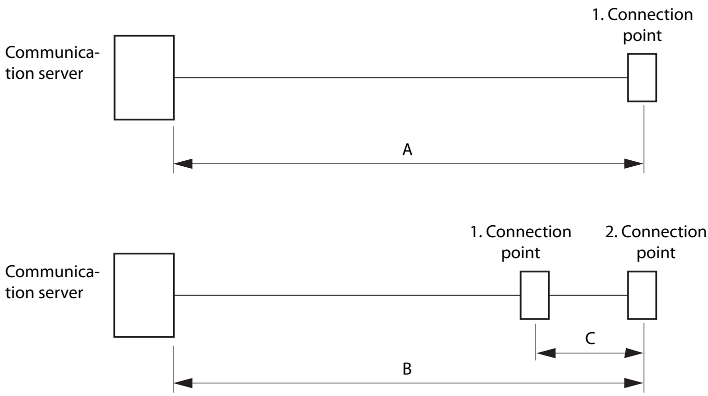


Fig. 46 DSI-AD2 bus

Notes

- The total length of the cables from the communication server to the system phone must not be less than 10 m.
- Only one system phone and only one phone of the Dialog 4200 series can be operated on each DSI-DASL interface. The maximum line length for a 0.5 mm wire diameter is set at 1000 m.

Restrictions

The maximum length of an DSI-AD2 bus is further restricted by:

- the maximum power requirements of the connected system phones and their supplementary equipment. In this context the SB-4+ and SB-8 DECT radio units are also considered as system phones.
- the line resistance (depending on the line length and wire diameter)

Tab. 61 Maximum power requirements of the system phones on the DSI bus

System phone 1)	Socket	Max. power input [mW]
MiVoice 5360 2)	DSI-AD2 interface	900
MiVoice 5361	DSI-AD2 interface	1220 3)
MiVoice 5370	DSI-AD2 interface	1220 3)
MiVoice 5380	DSI-AD2 interface	1340 3)
MiVoice 5370, MiVoice 5380 with power supply unit	DSI-AD2 interface	0
Expansion key module MiVoice M530	MiVoice 5370	300
Expansion key module MiVoice M530	MiVoice 5380	500
Expansion key module MiVoice M535	MiVoice 5370, MiVoice 5380	0 4)
Dialog 4220	DSI-DASL interface	500
Dialog 4222	DSI-DASL interface	660
Dialog 4223	DSI-DASL interface	680
EKP expansion key module	Dialog 4222, Dialog 4223	190
DECT radio unit without power supply unit SB-4+	DSI-AD2 interface	1700 5)
DECT radio unit without power supply unit SB-8	2 DSI-AD2 interfaces	1550 6)
DECT radio unit with power supply unit SB-4+/SB-8	1 or 2 DSI-AD2 interfaces	< 100

1) Assumptions:
System phones: In hands-free mode, loudspeaker on maximum volume, all LEDs lit
MiVoice 5380: Backlighting with maximum brightness
Expansion key modules: All LEDs lit
Radio units: Active call connection on all channels

2) Although no longer available, the phone is still supported.

3) The value can increase to approx. 600 mW if the power available at the DSI-AD2 bus allows it.

4) An MiVoice M535 always requires a power supply unit

5) The value applies to radio units with hardware version "-2". The value for hardware version "-1" is 300 mW lower.

6) The value applies to each interface and to radio units with hardware version "-2". The value per interface for radio units with hardware version "-1" is 150 mW lower.

The two diagrams below show the power available on the DSI-AD2 bus in relation to the line length and the wire diameter. The table can then be used to determine the number and type of system phones that can be connected to the DSI-AD2 bus under the given conditions. The power available can be calculated by measuring the loop resistance where the wire diameter is known.

Due to the different hardware versions of the radio units, the power available on the DSI-AD2 bus is not the same in every case:

Power available case A:

- Applies to all the system phones of the MiVoice 5300 series.
- Applies to the SB-4+/SB-8 DECT radio units with hardware version "-1".

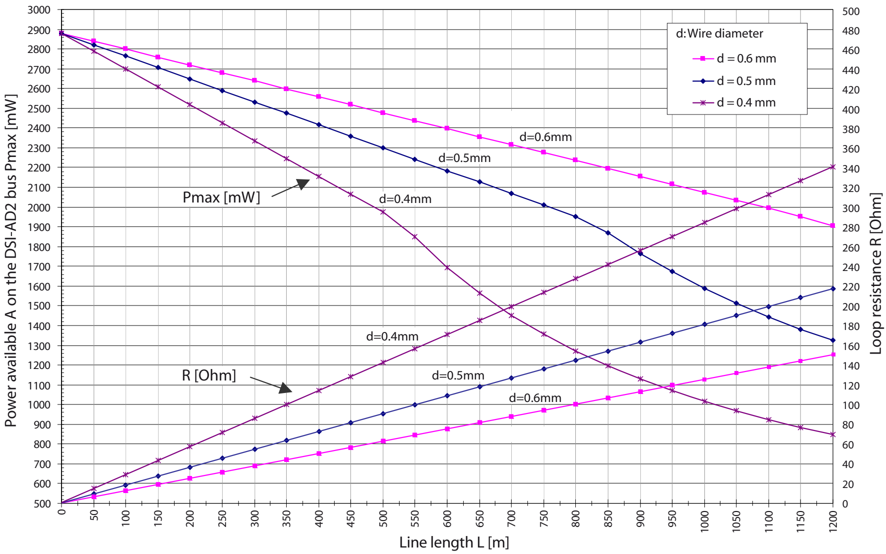


Fig. 47 Power available case A on the DSI-AD2 bus

Power available case B:

Applies to the SB-4+/SB-8 DECT radio units with hardware version "-2" and system phones of the Dialog 4200 series.

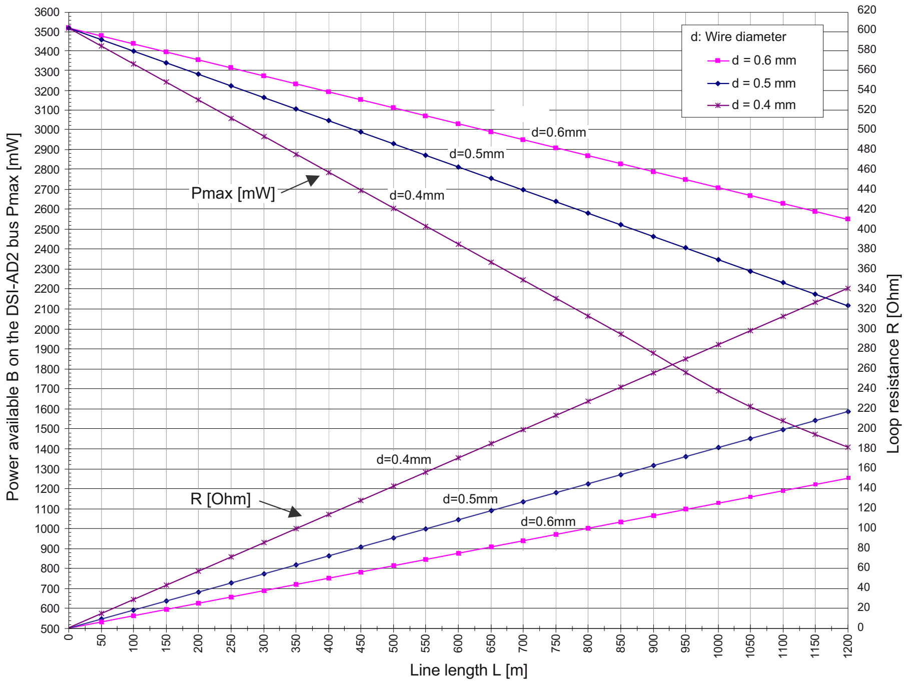


Fig. 48 Power available case B on the DSI-AD2 bus

Notes

- If another system phone is operated on the DSI-AD2 bus in addition to an MiVoice 5361, MiVoice 5370 or MiVoice 5380, at least one phone must be powered by a local power supply unit.
- An MiVoice 5370 or MiVoice 5380 with an MiVoice M535 expansion key module always requires a power supply unit.
- An MiVoice 5380 with 3 MiVoice M530 expansion key modules always requires a power supply unit. With 2 expansion key modules the use of power supply unit depends on the line length and the line cross-section.

Automatic detection of critical power supply situations

Only MiVoice 5360:

When a system phone (or a second such phone) is connected to the DSI bus, the maximum power input is automatically determined; all the system phones (incl. expansion key module and alphanumerical keyboard) connected to the interface are taken into account. The maximum power available is also determined based on the calculated line length (assumption: Diameter = 0.5 mm). If the calculated power available is below the maximum possible power input of the connected system phones, the message

Power supply critical xy m is generated on the phones connected last (accuracy approx.150 m).

System phones MiVoice 5361, MiVoice 5370 and MiVoice 5380 only:
During start-up, these system phones carry out a detailed measurement of the available power. A warning is shown on the display if the result is inadequate: *Line power too weak: External power supply required!*

Notes

– Depending on the power available based on the line length on the DSI-AD2 bus the ringing and hands-free volume decreases accordingly.
– The backlighting of the MiVoice 5380 display is brighter if the phone is powered by a power supply unit.

Rating examples

Example 1:

MiVoice 5370
Maximum power requirements as per Tab. 61: 1220 mW

Fig. 47 indicates:

- Maximum line length for a wire diameter of 0,4 mm: 840 m
- Maximum line length for a wire diameter of 0,5 mm: 1200 m
- Maximum line length for a wire diameter of 0,6 mm: 1200 m

Example 2:

An MiVoice 5380 with 2 MiVoice M530 expansion key modules
Power requirements as per Tab. 61: 1340 + 300 + 300 = 1940 mW.

Fig. 47 indicates:

- Maximum line length for a wire diameter of 0,4 mm: 520 m
- Maximum line length for a wire diameter of 0,5 mm: 820 m
- Maximum line length for a wire diameter of 0,6 mm: 1170 m

Example 3:

Evaluation of an existing line installation
Line diameter: 0.5 mm
Loop resistance: 120 Ω

Fig. 47 indicates:

- Line length: 660 m
- Power available: 2120 mW

Cable Requirements

Tab. 62 Requirements for an DSI bus cable

Core pairs × cores	1 × 2 o 1 × 4
Stranded	yes[1)]
Wire diameter, core	0.4…0.6 mm
Screening	recommended
Characteristic impedance	< 130 Ω (1 MHz)

1) Note: max. 25 m can be crossed unstranded.
(CH: Applies also to cable type G51)

Installation rules

- If an Mitel DECT radio unit is used, do not connect any other system phone to the same DSI bus.
- If *Interface type* is set to *DSI-DASL*, only connect one system phone or one phone of the Dialog 4200 series to the DSI bus.
- Do not use any terminating resistors at the bus extremity.
- Avoid using different cable cross-sections on the same bus
- Use the supplied cables for connecting the system phones
- Cabling of AD2 terminals is restricted to pairs of a separate dedicated cable(s).[1)].

Terminals

The following system terminals can be operated on the DSI-AD2 bus:

- MiVoice 5300 series system phones
- Mitel DECT radio units

The system phones on an DSI-AD2 bus are addressed via a single-digit terminal selection digit (TSD).

Example:
The address of a system phone with TSD 2 on DSI interface 3.5 is 3.5-2.

Only system phones of the Dialog 4200 series can be operated on a DSI-DASL bus.

1) Applies in Australia only

4. 7. 3. 2 BRI-S terminal interfaces

Fitting the corresponding interface cards means that BRI-S terminal interfaces are available on the RJ45-sockets on the front panel of the cards. The possible RJ45 sockets are highlighted in colour in the figure below.

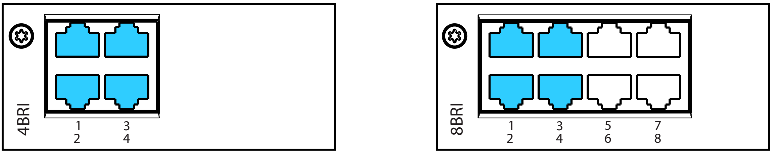
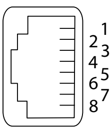
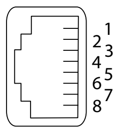


Fig. 49 Connection possibilities for BRI-S terminal interfaces

Note
With the 8BRI card, only the interfaces of sockets 1 to 4 are available for BRI-S terminal interfaces. The interfaces of sockets 5 to 8 are permanently configured to BRI-T.

Connection

Tab. 63 Connection of BRI-S terminal interfaces

Communication server			Cable cores	Connection socket		
Socket	Pin	BRI-S signal		BRI-S signal	Pin	Socket
	1	–		–	1	
	2	–		–	2	
	3	c	←	c	3	
	4	f	→	f	4	
	5	e	→	e	5	
	6	d	←	d	6	
	7	–		–	7	
	8	–		–	8	

S bus configuration

The S bus is a four-wire, serial ISDN bus based on the DSS1 protocol (ETSI standard). It starts in each case at an BRI-S interface of the communication server. Four bus configurations are possible, depending on the line length and the number of terminals:

Tab. 64 S bus configurations depending on line length and the number of terminals.

S bus	Short	Short, V-shaped	Long	Point-to-point
Length (max.) Server ↔ terminal Terminal 1 ↔ Terminal 4	 150 m –	 2 × 150 m –	 500 m 20 m	 1’000 m –
Number of terminals (max.)	8	8	4	1

Note

The maximum number of terminals per S bus depends on the power requirements of the terminals (see "Restrictions", page 146).

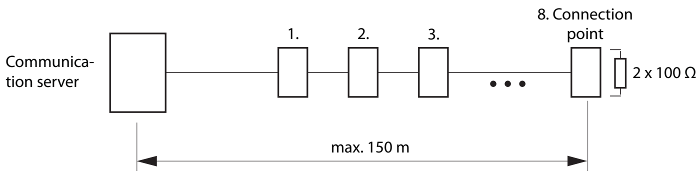


Fig. 50 S bus, short

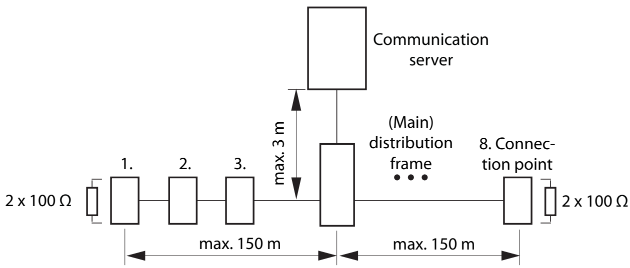


Fig. 51 S bus, short, V-shaped

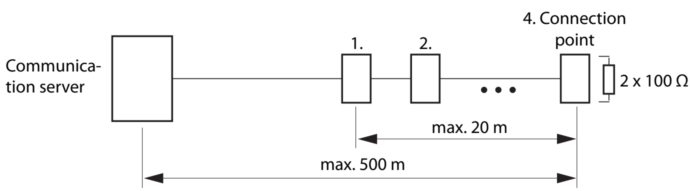


Fig. 52 S bus, long

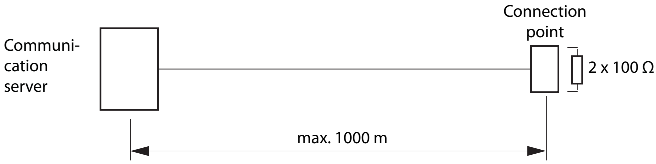


Fig. 53 S bus, point-to-point

Greater distances (up to 8km) can be achieved using a standard commercial S bus extension.

Restrictions

The maximum number of terminals per S bus is further restricted by the power requirements of the terminals and their supplementary equipment:

Tab. 65 Power balance on the S bus

	Power available [W]
S bus short	5[1)]
S bus, long	3.5[1)]

1) These values are based on a wire diameter of 0.5 mm.

The number of terminals is the sum of the power requirements of the individual terminals and the power available on the S bus.

Connection sockets

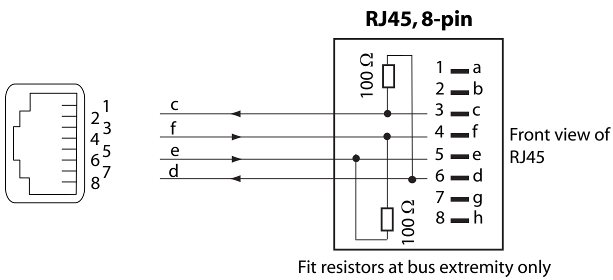


Fig. 54 RJ45 connection, single socket

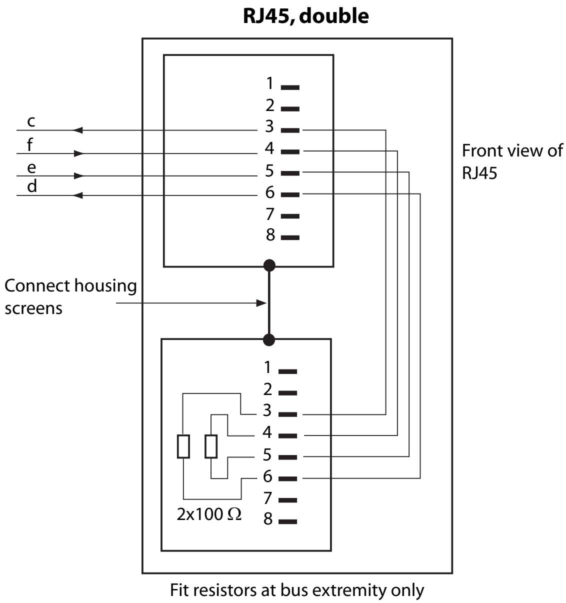


Fig. 55 RJ45 connection, double socket

Installation rules

Always terminate the bus extremity with 2 × 100 Ω (0.25 W, 5%)!

Note
Circuit type as per EN/IEC 60950: SELV

Cable Requirements

Tab. 66 Requirements for an S bus cable

Core pairs × cores Stranded Wire diameter, core Screening	1 × 4 or 2 × 2 yes 0.4…0.6 mm recommended
Ohmic resistance Characteristic impedance Wave attenuation Near/crosstalk attenuation	< 98 Ω/km (conductor), < 196 Ω/km (loop) < 125 Ω (100 kHz), < 115 Ω (1 MHz) < 6 dB/km (100 kHz), < 26 dB/km (1 MHz) > 54 dB/100 m (1 kHz to 1 MHz)

Terminals

The ETSI protocol must be set in the interface configuration.

Up to 8 terminals of different types can be connected to one S bus.

- Standard ISDN terminals
- ISDN Terminal Adapter
- PC with ISDN card
- Group 4 fax machines[1)], etc.

Two call connections are possible simultaneously for each S bus.

4. 7. 3. 3 FXS terminal interfaces

The call manager card CPU1 already contains 4 FXS terminal interfaces, which are routed through to the front panel of the card and labelled accordingly. The number of available FXS terminal interfaces can be increased by fitting interface cards. The RJ45 connector assignment is identical. The possible RJ45 sockets are highlighted in colour in the figure below.

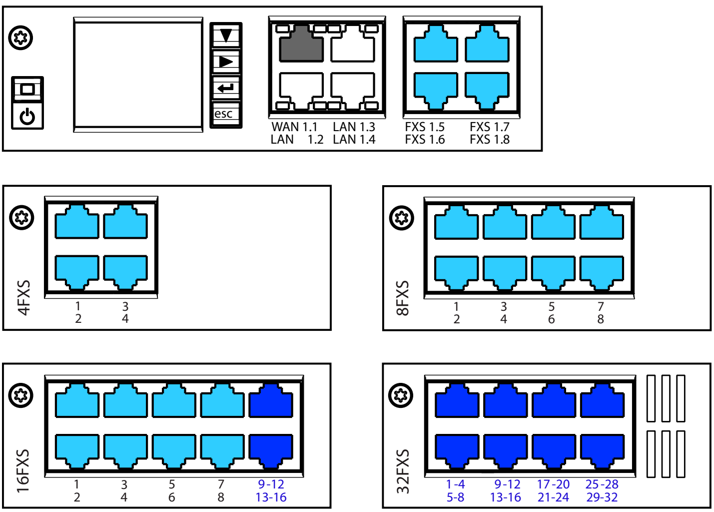


Fig. 56 Connection possibilities for FXS terminal interfaces

On terminal cards with 16 or more interfaces some or all of the RJ45 sockets are multiply assigned. The signals can be split again to individual RJ45 sockets using patch ca-

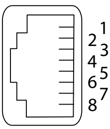
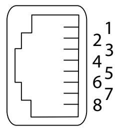
1) Not possible within an AIN

bles and the fan-out panel FOP (see "Fan-out panel FOP", page 157) or with 8-fold assigned connecting cables (see e.g. "Prefabricated system cable 4 x RJ45", page 117).

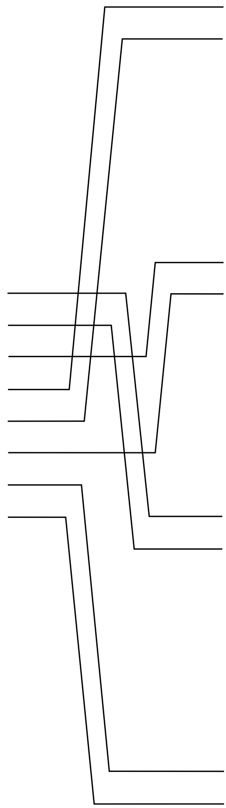
Multiply assigned RJ45 sockets are colour-coded in blue.

Connection

Tab. 67 Connection of individually assigned FXS terminal interface

Communication server			Cable cores	Connection socket		
Socket	**Pin**	**Analogue signal**		**Analogue signal**	**Pin**	**Socket**
	1	–		–	1	
	2	–		–	2	
	3	–		–	3	
	4	a		a	4	
	5	b		b	5	
	6	–		–	6	
	7	–		–	7	
	8	–		–	8	

Tab. 68 Connection of four-fold assigned FXS terminal interface

Communication server			Splitting with fan-out panel FOP or 8-fold assigned connecting cables	Connection socket		
Socket	Pin	Analogue signal		Analogue signal	Pin	Socket
				–	1	
				–	2	
				–	3	
				1a	4	
				1b	5	
				–	6	
				–	7	
				–	8	
				–	1	
				–	2	
				–	3	
				2a	4	
	1	3a		2b	5	
	2	3b		–	6	
	3	2a		–	7	
	4	1a		–	8	
	5	1b		–	1	
	6	2b		–	2	
	7	4a		–	3	
	8	4b		3a	4	
				3b	5	
				–	6	
				–	7	
				–	8	
				–	1	
				–	2	
				–	3	
				4a	4	
				4b	5	
				–	6	
				–	7	
				–	8	

Multifunctional FXS interfaces

The analogue interfaces of the FX cards are multifunctional. Depending on the terminal or function they are configured individually in the *Interface configuration* using and switched over internally accordingly.

Tab. 69 Mode of the FXS interfaces

FXS mode	**Socket**
Phone/fax	Analogue DTMF and pulse dialling terminals such as phones, fax, modem, answering machines, etc.
2-wire door	Analogue two-wire door intercom
External audio source	Audio interface for connecting playback equipment with line output.
Control output	Ports for switching external equipment.
Control input	Ports for switching internal switch groups.
General bell	Commercial auxiliary bells

After a first start all the FXS interfaces are configured on *Phone / Fax*.

CAUTION!
Terminals connected to FXS interfaces can be damaged if the configuration of the FXS interface mode is unsuitable.

Note
Circuit type as per EN/IEC 60950: TNV-2

FXS mode: Phone/fax

In this mode the following analogue terminals can be connected:

- Analogue phones with DTMF or pulse dialling (earth key is not supported)
- Radio units for cordless phones
- Group 3 fax[1)]
- Answering machines
- Modem

1) Transmission with the T.38 protocol is recommended for Fax over IP. The corresponding media resources need to be allocated.

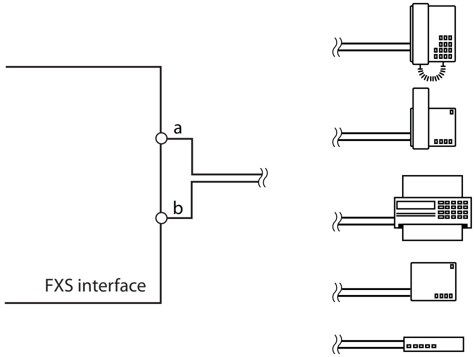


Fig. 57 Connection for FXS mode: Phone/fax

Parameter	Interface FXS1...FXS6 of Mitel SMB Controller
OPS Loss Plan Support	No
DNIC Support	No
LS Trunk Support	Yes
LS Class Support	Yes
GS Trunk Support	No
REN per line (ONS)	2
Open circuit Ringing Voltage (VRMS)	Nominal = 70V (trapezoidal balance) or 40V rms (sinus wave) Note: This setting is country dependent
ONS Loop Length (Miles)	600 Ohm Loop (with 300 Ohm set) 3.45 (22AWG) 2.17 (24AWG) 1.33 (26AWG)
ONS Line Feed Voltage	30V for normal 56V for long loop
ONS WMI Lamp Strike Voltage (VDC)	The following formats are supported: - High voltage (>90Vdc) Note: This setting is country dependent - Low voltage LED type: 12Vrms at 20Hz for 100ms - FSK - Polarity reversal

Fig. 58 Specification for FXS mode: Phone/fax (used for USA/Canada only)

Ports 1.5 and 1.6 on the call manager card and in each case the first two ports of FXS cards (X.1 and X.2) are designed for long lines. The no-load voltage at these ports is 51 VDC. All the other ports have a no-load voltage of 30 VDC. The loop current is limited to 25 mA on all ports.

Tab. 70 Cable requirements for FXS mode: Phone/fax

	Ports for long lines	**Normal ports**
Core pairs × cores	1 × 2	1 × 2
Stranded	only with lengths > 200 m	only with lengths > 200 m
Wire diameter, core	0.4 … 0.8 mm	0.4 … 0.8 mm

	Ports for long lines	Normal ports
FXS resistance	max. 2 × 625 Ω	max. 2 × 250 Ω
Line length 0.6 mm diameter	max. 10 km	max. 4 km
Screening	not required	not required

Parameter	FXS Interface Mitel 470
OPS Loss Plan Support	No
DNIC Support	No
LS Trunk Support	Yes
LS Class Support	Yes
GS Trunk Support	No
REN per line (ONS)	2
Open circuit Ringing Voltage (VRMS)	Nominal = 52V (trapezoidal balance)
ONS Loop Length (Miles)*	600 Ohm Loop (with 300 Ohm set) 3.45 (22AWG) 2.17 (24AWG) 1.33 (26AWG)
ONS Line Feed Voltage	30V for normal 56V for long loop
ONS WMI Lamp Strike Voltage (VDC)	High voltage (>90Vdc) is not supported. The following formats are supported: - Low voltage LED type: 12Vrms at 20Hz for 100ms - FSK - Polarity reversal

* Loop length is limited by minimum 40Vrms ringing voltage at telephone set with 2 REN ringer load

Fig. 59 Specification for FXS mode: Phone/fax (used for USA/Canada only)

FXS mode: 2-wire door

In this mode 2-wire door intercoms with DTMF control functions can be connected. The no-load voltage in this mode is 24 VDC. The loop current is limited to 25 mA.

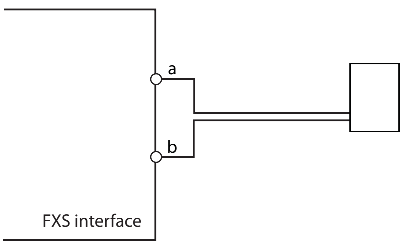


Fig. 60 Connection for FXS mode: 2-wire door

Tab. 71 Cable requirements for FXS mode: 2-wire door

Core pairs × cores	1 × 2
Stranded	only with lengths > 200 m
Wire diameter, core	0.4 … 0.8 mm
FXS resistance	max. 2 × 200 Ω
Line length 0.6 mm diameter	max. 3 km
Screening	not required

FXS mode: External audio source

One FXS interface per communication server can be configured for connecting an audio device. In this mode the FXS interface becomes an audio input that can be used for the following purposes:

- to play music or an announcement to connections with callers on hold ("Music on hold" function).
- to play music or an announcement for the announcement service (announcement prior to answering), voice mail greetings or for "Music on hold" and then to store it as a wave file.

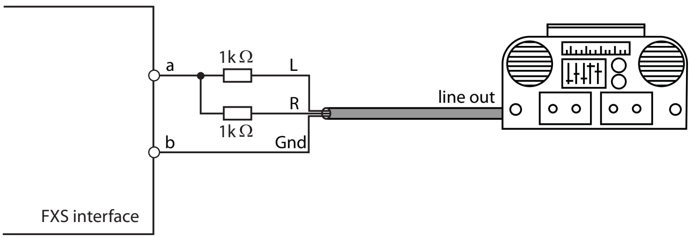


Fig. 61 Connection for FXS mode: External audio source

Any playback equipment (tape recorder, CD player, etc.) with a line output can be used as the audio source. It is advisable to merge the left/right audio signal via 2 resistors (see Fig. 61).

CAUTION!
The default value of all FXS interfaces is set to *Phone/Fax*. Audio equipment may be damaged by the DC or AC voltage imposed.
Make sure that the mode of the FXS interface is configured to External audio source before connecting audio equipment.

Note:
The customer is responsible for all copyright matters relating to any music playback.

Tab. 72 Technical data for FXS mode: External audio source

Input impedance	approx. 15 kΩ
Input level	configurable
Input circuit	asymmetrical
Output resistance audio source	< 1 kΩ
Installation cable	NF cable screened (required for low levels)

FXS mode: Control output

If an FXS interface is configured as a control output, the signal can be used to control external devices or equipment (e.g. heating, alarm or outdoor lighting systems).

The no-load voltage is 24 VDC; the current is limited to 25 mA. A connected relay must be of the type 24 VDC and must not draw more than 300 mW in power.

There are no special requirements for the cables.

CAUTION!
Control outputs must have a floating connection.

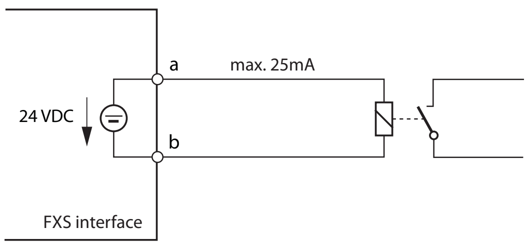


Fig. 62 Connection for FXS mode: Control output

FXS mode: Control input

If FXS interfaces are configured as control inputs, one or more of the switch groups can be switched between Positions 1, 2 and 3. An external switch or a relay is connected for this purpose. An LED can be connected to the circuit to indicate the switch state. The no-load voltage is 24 VDC; the current is limited to 25mA.

The permissible switch and loop resistances are as follows:

- Active state (On): < 1 kΩ
- Passive state (Off): > 4 kΩ

There are no special requirements for the cables.

CAUTION!
Control inputs must have a floating connection.

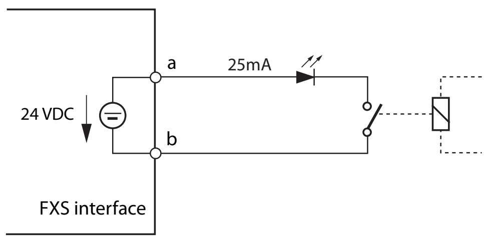


Fig. 63 Connection for FXS mode: Control input

In the switch group configuration in (=*xb*) the ports are assigned to the control inputs of a switch group. To be able to control all 3 switch positions of a switch group, you need 2 control inputs which switch the switch position of the switch group depending on the status.

Tab. 73 Switch group control via the control inputs

FXS control input 1	*FXS control input 2*	**Switch positions of the switch group**
Off	Off	Position 1
On	Off	Position 2
any	On	Position 3

Other conditions:

- The same control inputs can control one or more switch groups.
- The same switch group can only be switched by the 2 assigned control inputs.
- Control of the switch groups using the control inputs takes priority over control using function codes.

FXS mode: General bell

One FXS interface per communication server can be configured for the connection of a general bell. It is possible to use commercial auxiliary bells designed for connection in parallel to analogue terminals as a general bell. However the impedance of the connected general bell (or total impedance in the case of several devices connected in parallel) must not fall below 1 kΩ. The ringing voltage is 48 VAC. A 48 V relay must be interposed when connecting a large number of auxiliary bells.

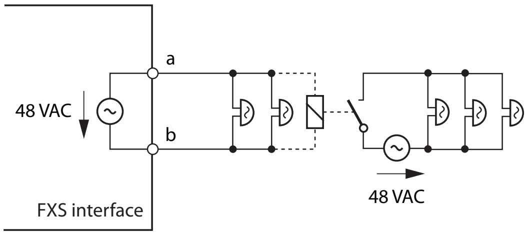


Fig. 64 Connection for FXS mode: General bell

See also
"General bell on FXS interface" in the "System Functions and Features" System Manual.

4. 7. 4 Fan-out panel FOP

All interface cards with 16 or more interfaces have four-fold assigned RJ45 sockets. With the fan-out panel FOP a total of 10 four-fold assigned RJ45 sockets can be split to individual RJ45 sockets.

The fan-out panel (FOP) takes up the space of one height unit in a rack and can be fitted directly above or below the communication server.

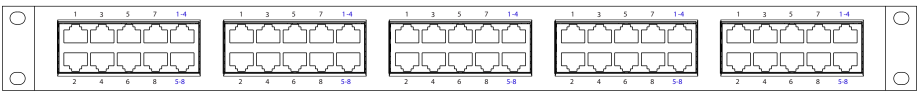


Fig. 65 Front panel, FOP fan-out-panel

Fan-out panels can also be offset, e.g. as floor distributors.

Note:
The fan-out panel FOP must be installed in a 19” rack.

Connection

The diagram below shows the connection of an interface card 16DSI with terminals. This card has 2 four-fold assigned RJ45 sockets. The 8 individually assigned RJ45 sockets are connected directly while the 2 four-fold assigned sockets are looped via the front panel of the fan-out-panel connector (FOP) strip using 2 patch cables.

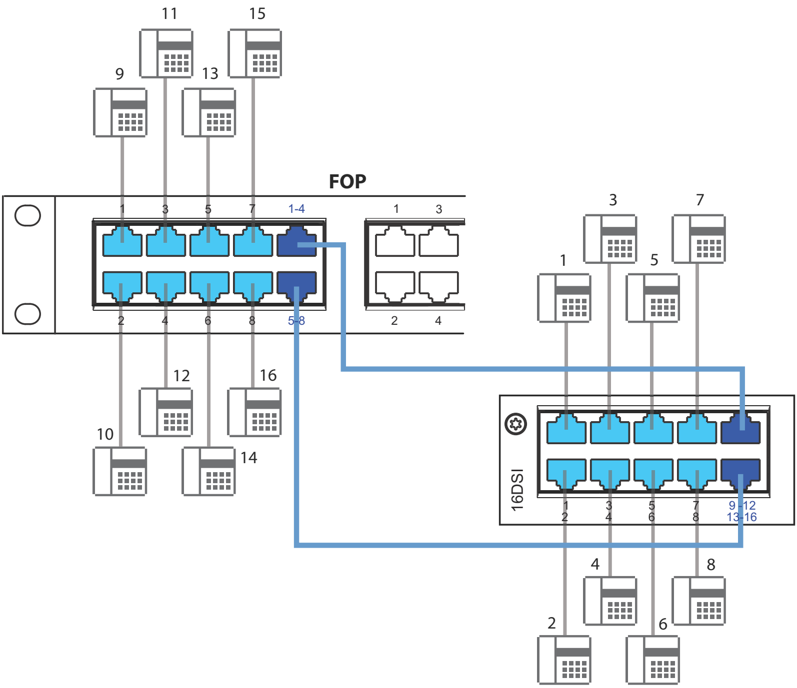


Fig. 66 Connection of four-fold assigned sockets via FOP connector strip

The patch cables are available separately in lengths of 1 and 2 m (see "Equipment Overview", page 258).

The internal wiring of the fan-out panel is shown in the table below. The wiring is shown for sockets 1 - 4. Sockets 5 - 8 are wired accordingly.

Tab. 74 Wiring of sockets 1–4 in the fan-out panel FOP

Fan-out panel FOP			Internal wiring	Fan-out panel FOP		
Socket	Pin	Signal		Signal	Pin	Socket
				–	1	1
				–	2	
				–	3	
				1a	4	
				1b	5	
				–	6	
				–	7	
				–	8	
				–	1	2
				–	2	
				–	3	
				2a	4	
1-4	1	3a		2b	5	
	2	3b		–	6	
	3	2a		–	7	
	4	1a		–	8	
	5	1b		–	1	3
	6	2b		–	2	
	7	4a		–	3	
	8	4b		3a	4	
				3b	5	
				–	6	
				–	7	
				–	8	
				–	1	4
				–	2	
				–	3	
				4a	4	
				4b	5	
				–	6	
				–	7	
				–	8	

Socket

The FOP fan-out-panel does not require a power supply.

4. 7. 5 Ethernet interfaces

The communication server Mitel 470 has a Gbit Ethernet switch on the call manager card. Three LAN interfaces are routed to the front panel of the call manager card and labelled accordingly. The RJ45 sockets are highlighted in colour in the figure below.

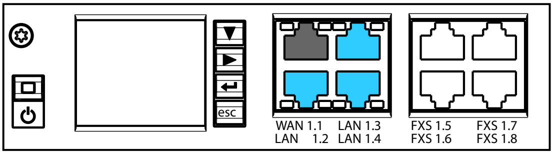
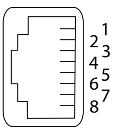


Fig. 67 Connection possibilities for Ethernet interfaces

Note
Circuit type as per EN/IEC 60950: SELV

Socket

Tab. 75 Connection of Ethernet interfaces

RJ45 socket	Pin	Signal
	1	TX D1+
	2	TX D1–
	3	RX D2+
	4	BI D3+
	5	BI D3–
	6	RX D2–
	7	BI D4+
	8	BI D4–

Settings

The IP address can either be taken from a DHCP server in the IP network or configured statically. If a DNS server is used, the communication server can also be addressed via its host name.

Tab. 76 Default values, IP address

Parameter	Parameter value
IP address	192.168.104.13
Subnet mask	255.255.255.0
Gateway	0.0.0.0
DHCP	*yes*
Host name	*<Model name>-<MAC-Address>*[1)] Example: Mitel430-00085d803100

1) This entry is hidden and does not appear in the parameter's input field

First-start response

The IP addressing after a first start depends on whether a static IP addressing is already stored from a previous configuration. A static IP addressing (IP address, subnet mask, gateway) entered manually is stored and remains available after a first start. This means that the communication server remains accessible via Ethernet interface in the same way as before the first start.

If no IP addressing is entered (e.g. after initial delivery), the communication server is started with DHCP after a first start. The communication server tries to log on with the DHCP server and to enter its host name on the DNS server. If log on is successful the communication server is accessible via the host name.

If the communication server cannot find a DHCP server within 90 seconds, it deactivates the DHCP mode and is then accessible via the standard IP address (see Tab. 76) with a direct connection.

Note:
DHCP is deactivated only temporarily and is reactivated after a subsequent restart.

Cable types

The Ethernet switch on the communications server features Auto MDI/MDIX. With the automatic detection straight or crossover LAN cables can be used for all connection types.

Configuration

The Ethernet interfaces routed to the front panel can be configured individually in the *IP addressing* (=9g) view. In addition to Auto modes, manual settings are also possible for *Speed* and *MDI type*.

Status LED

The status of the Ethernet interfaces is indicated by the green and yellow LEDs directly on the interface in question.

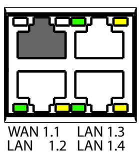


Fig. 68 Status LED on the Ethernet interfaces

Tab. 77 Status LED on the Ethernet interfaces

Green LED	Yellow LED	Speed	State
On	On	10 Mbit/s	Port has a connection with the network
Flashing	Flashing	10 Mbit/s	Port is receiving or sending data
On	Off	100 Mbit/s	Port has a connection with the network
Flashing	Off	100 Mbit/s	Port is receiving or sending data
Off	On	1 Gbit/s	Port has a connection with the network
Off	Flashing	1 Gbit/s	Port is receiving or sending data

Cable Requirements

Use commercial Cat. 5 cable, or choose a cable type with the following characteristics:

Tab. 78 Requirements for an Ethernet cable

Core pairs × cores	4 × 2
Stranded	yes
Wire diameter, core	0.4...0.6 mm
Screening	yes
Category	Cat. 5 minimum

See also:
For more information about the Ethernet interface on the application card, see the CPU2-S application card installation manual.

4. 8 Installing, powering, connecting and registering terminals

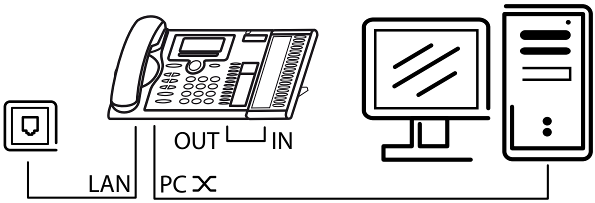
4. 8. 1 IP system phones

Accesses

Tab. 79 Socket connections of the IP system phones of the MiVoice 5300 IP series

LAN	PoE Ethernet interface for connection to the IP network
PC ⤫	Socket connection for a workstation PC (integrated 100Base-T switch, available on MiVoice 5370 IP and MiVoice 5380 IP)
	Handset socket
	Headset socket

	Power supply socket for connecting a power supply if PoE is not available
OUT 	Connect expansion key module MiVoice M530/MiVoice M535 (available on MiVoice 5370 IP and MiVoice 5380 IP)



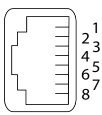
Integrated switch (MiVoice 5370 IP and MiVoice 5380 IP)

You can use the integrated 100Base-T mini-switch to connect other network terminals (e.g. PC, printer), thereby reducing the amount of cabling required.

Power supply

If your network supports Power-over-Ethernet, the IP system phone is powered directly via the LAN connection and there is no need to connect the power supply available as an option.

Tab. 80 Power over Ethernet

RJ45 socket	Pin	Signal	PoE power supply (Variant 1)	PoE power supply (Variant 2)
	1	Rx	DC+	—
	2	Rx	DC+	—
	3	Tx	DC-	—
	4	—	—	DC+
	5	—	—	DC+
	6	Tx	DC-	—
	7	—	—	DC-
	8	—	—	DC-

Depending on the power requirements different classes are defined in the IEEE 802.3af standard. The following table provides information on the class allocation of the IP system phones.

Tab. 81 PoE class allocation

Class	Max. load, PSE1)	Max. power requirement, PD2)	IP system phones
1	4.0 W	0.44...3.84 W	MiVoice 5360 IP, MiVoice 5361 IP
2	7.0 W	3.84...6.49 W	MiVoice 5370 IP3), MiVoice 5380 IP4)
3	15.4 W	6.49...12.95 W	

1) PSE (Power Source Equipment) = power supply device, e.g. a switch

2) PD (Powered Device) = power consumer, e.g. an IP system phone

3) including an MiVoice M530 or MiVoice M535 expansion keypad

4) including up to three MiVoice M530 or MiVoice M535 expansion keypads

You can obtain information on how to operate and register the IP system phones on a MiVoice Office 400 communication server in the WebAdmin online help.

4. 8. 2 Mitel 6800/6900 SIP phone series

Mitel SIP phones are platform-independent phones with a wide range of features. They can also be perfectly integrated into one of the Mitel Platforms and used as a system phone. Mitel SIP Phones on MiVoice Office 400 first support MiVoice Office 400 features and have a separate user's guide. Many of the device-specific functions are less significant or are not used at all. Please read the Mitel SIP administration instructions if you wish to carry use device-specific functions or carry out device-specific settings. Device-specific installation instructions are available for installing the phones. You can obtain information on how to register a Mitel SIP phone on a MiVoice Office 400 communication server in the WebAdmin online help.

4. 8. 3 Standard SIP phones and standard SIP terminals

For information on installation, powering and connection, please refer to the installation instructions of the corresponding phones and terminals. Information on how to register Mitel or third-party standard SIP phones and standard SIP terminals as internal users in MiVoice Office 400 is given in WebAdmin.

4. 8. 4 Mobile/external phones

The integration of mobile/external phones in the MiVoice Office 400 communication system is described in the System Manual "System Functions and Features".

4. 8. 5 OIP and other applications

Mitel Open Interfaces Platform (OIP) is also available as OIP Virtual Appliance and can be installed on the same server as the Virtual Appliance communication server. The operating requirements and installation instructions for the OIP applications

MiVoice 1560 PC Operator and Mitel OfficeSuite are described in the "Mitel Open Interfaces Platform" System Manual.

4. 8. 6 Digital system phones

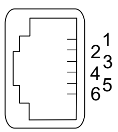
4. 8. 6. 1 General information

Accesses

The connections on the underside of the system phones are identified by the symbols. The meaning of the symbols is described in the corresponding operating instructions.

DSI terminal interface

Tab. 82 DSI interface on the phone

RJ45 socket	Pin	Signal
	1	—
	2	—
	3	b
	4	a
	5	—
	6	—

Note:
The total length of the cables from the communication server to the system phone must not be less than 10 m.

Terminal selection

Two system phones can be connected to a DSI interface (DSI-AD2 only). The system can only differentiate the two system phones by the position of the address switch on the phone. The following settings are possible (TSD = Terminal Selection Digit):

- TSD1
- TSD2

Note:
In the following cases *Not Configured* is displayed along with the node number, the slot number and the port number. In this state the system phone is not ready for operation:

- A terminal has been created at the connected port, but the address selection switch is incorrectly set.
- No terminal has yet been created at the connected port.

User allocation

In the configuration each terminal is assigned to a user or a free seating pool. If a terminal has been created at the connected port and the address selection switch is correctly selected but no user or free seating pool is allocated to the terminal, the system phone display reads *No Number* and indicates the terminal ID. In this state the system phone is not ready for operation.

Terminal type

The terminal type is specified along with the configuration of the system. Lines are also assigned to the line keys there.

Note:
If the terminal type configured is incorrect, the system phone display shows the warning *Wrong phone type*. In this situation, although the system phone can be used for basic telephone operations, none of the added features will be available. The terminal type must be entered via WebAdmin or on the terminal via login to the system configuration.
Carrying out a logon on the system phone: Long keypress (long click) on a function key. *Set new phone type* appears next. Confirm with Foxkey *Yes*.

4. 8. 6. 2 MiVoice 5361 / 5370 / 5380

These IP system phones can be both desktop-mounted and wall-mounted.

Mounting the phone

The following points are described in detail in the User's Guides for MiVoice 5361 / 5370 / 5380:

- Set-up as a desk phone (choice of two different set-up angles)
- Wall mounting
- Connecting one or more MiVoice M530 or MiVoice M535 expansion key modules.
- Connection of a headset to DHSG standard.

Note:
To prevent any damage to the phone, always disconnect the phone from the power supply first before connecting a headset to DHSG standard.

Mounting the Bluetooth module

The MiVoice 5380 can also be equipped with a Bluetooth module as an option. To install (see Fig. 69), proceed as follows:

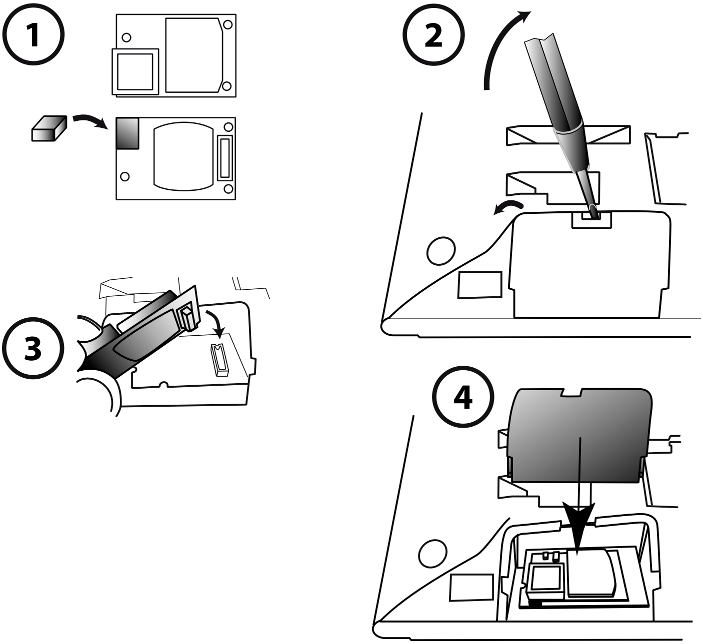


Fig. 69 Assembly of the Bluetooth module

⚠ **CAUTION!**
The system's reliability can be adversely affected by electrostatic discharges caused by touching electronic components and elements, and subsequent damage can result. Always observe the ESD guidelines.

1. Fit foam spacers on the connector side of the Bluetooth module (for the position of the foam spacer see ①). The spacer ensures that the Bluetooth module sits securely.
2. Carefully remove the cover for the Bluetooth module on the underside of the phone using a suitable screwdriver (see ②).
3. Connect the Bluetooth module. Make sure it is securely fitted (see ③).
4. Fit the cover for the Bluetooth module back into place and press home until it snaps into place (see ④).

Powering the phone

The MiVoice 5360, MiVoice 5361 MiVoice 5370 and MiVoice 5380 system phones are normally powered via the DSI bus. However there are several reasons that require powering with a plug-in power supply:

- Long line

- 2 phones on the same bus
- 1 or more expansion key modules on the phone
- Terminal power supply of the communication server is overloaded

Only use the corresponding plug-in power supply unit with FCC connector available as an option. It is connected either to the phone itself or, when using one or more expansion key modules, on the last expansion key module.

See also
The power available on the DSI bus depending on the line length and the wire diameter, and the power input of the system phones are described in the chapter "DSI terminal interfaces", page 135 ff.

Connecting the phone

1. Setting the DSI bus address on the system phone's underside:
   – TSD1 = address switch on position 1
   – TSD2 = address switch on position 2
2. Plug the connector into the socket-outlet.
3. If the system is configured, test the operation of the system phone.
4. Label the phone as indicated in the operating instructions.

4. 8. 7 DECT radio units and cordless phones

The locations determined for the cordless phones, charging bays and radio units during the planning phase need to be checked against the following criteria:

- Influence on radio operation
- Ambient conditions

Influences on radio operation

Radio operation is affected by the following influences:

- Outside interference (EMC)
- Obstacles in the surrounding area affect the radio characteristic

To achieve optimum conditions for radio operation, observe the following points:

- Optimum radio operation depends on the radio unit → cordless phone line of sight.
- Walls act as an obstacle to the propagation of radio waves. Losses depend on the wall thickness, construction material and reinforcement used.

- Do not place radio units and cordless phones in the immediate vicinity of TV sets, radios, CD players or power installations (for reasons of EMC, e.g.distribution boxes, rising power lines).
- Do not place radio units and cordless phones near X-ray installations (EMC).
- Do not place radio units and cordless phones near metal partitions.
- Observe the minimum distance requirements between adjacent radio units (see Fig. 71).
- Minimum distance between cordless phones for fault-free operation: 0.2 m. (The charging bays of the Office 135 can be linked using connecting strips. However, operating several phones on interconnected charging bays can lead to malfunctions.)
- Minimum distance between charging bays with cordless phones on-hook for fault-free operation: 0.2 m.

Ambient conditions

- When installing: Ensure convection (space for ventilation).
- Avoid excessive dust.
- Avoid exposure to chemicals.
- Avoid direct sunlight.
- See also technical data in Tab. 123.

Note:
If these requirements cannot be met (e.g.outdoor installation), use the appropriate protective housing.

4. 8. 7. 1 Installing the radio units

Do **not** remove the cover of the radio unit. (Warranty protection will lapse if the cover is removed)

Fit the mounting bracket (see Fig. 70 dimensional drawing for wall mounting). Observe the minimum distances (see Fig. 71).

Position the DSI socket(s) near the radio unit.
Each radio unit requires one DSI bus (two optional on the SB-8): Do not connect any other terminals.

The radio units can be powered from the communication server up to the maximum line length of 1200 m specified for operation (wire diameter 0.5 mm). The plug-in power supply unit for is the same as the one for the Office 135 charging bay.

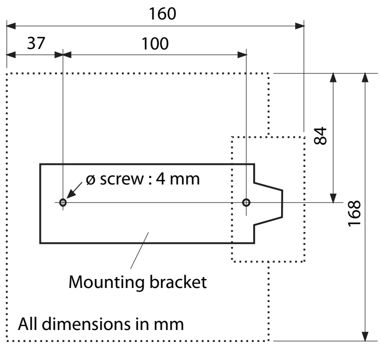
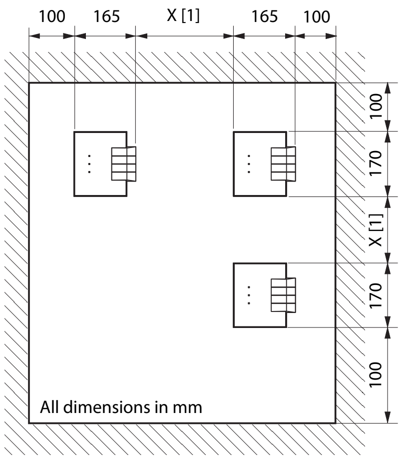


Fig. 70 Dimensional drawing for wall-mounting the mounting bracket



[1] X = 200: Minimum distance if the radio units are connected to the same communication server (synchronous)
X = 2000:Minimum distance if the radio units are not connected to the same communication server (not synchronous)
Make sure the minimum distances are observed

Fig. 71 Installation distances

Connecting the radio unit

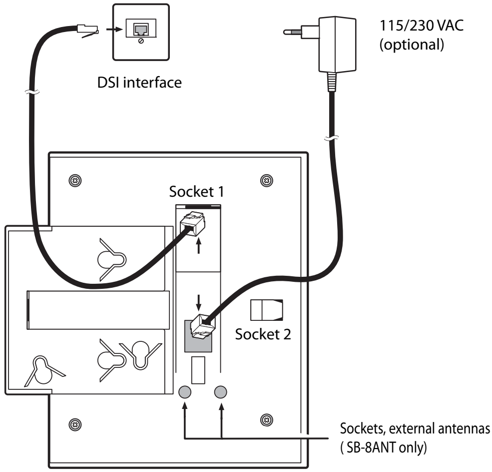
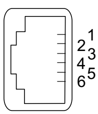


Fig. 72 Underside of the radio units with connection points

Tab. 83 Connections on the Mitel DECT radio units

RJ12 sockets	Pin	Socket 1: DSI interface		Socket 2: Power supply
		SB-4+	SB-8 / SB-8ANT	SB-4+ / SB-8 / SB-8ANT
	1	Local power supply –	Local power supply –	Local power supply –
	2	—	b2	—
	3	b1	b1	—
	4	a1	a1	—
	5	—	a2	—
	6	Local power supply +	Local power supply +	Local power supply +

If an SB-8 / SB-8ANT is operated on two DSI interfaces, it is recommended always to use two neighbouring ports.

Mitel Advanced Intelligent Network:

As the DECT systems of the individual nodes in an AIN do not run synchronously, the two DSI interfaces of an SB-8 / SB-8ANT must always be connected to the same node.

Tab. 84 Operating state display on Mitel DECT radio units

LED flashing (two LEDs on the SB-8)	Information
green	Operating state
red / green	Startup procedure running

LED flashing (two LEDs on the SB-8)	Information
orange	Transmission of DECT sequences
red	Fault
not flashing and not lit	LED switched off or radio unit defective or not in operation

For further display variants, see "Operating state of the Mitel DECT radio units", page 250

4. 8. 8 Analogue phones Mitel 6710 Analogue, Mitel 6730 Analogue

The phones can be used as desktop model or as wall model.

Connecting the phone

1. Stick the connector on the longer, straight end of the handset cord on the underside of the phone into the socket with the handset symbol until it snaps into place. Feed the cable through the strain relief and connect the other end to the handset.
2. Feed the small connector of the phone connection cable on the rear side of the phone into the socket until it snaps into place. Stick the connector on the other end into the phone cord.

Preparing the phone for message waiting indication (MWI)

The phone can detect various types of notifications (polarity reversal, high voltage and frequency shift keying (FSK). The notification type is set with the MWI switch on the underside of the phone. "0" = Off, "HV" = High voltage, "−/+" = Polarity reversal. The notification type Frequency reversal (FSK) is always active, regardless of the switch setting (Mitel 6730 Analogue only).

The MiVoice Office 400 communication servers support the following notification types (Parameter *MWI mode* configurable for each FXS Interface separately):

Tab. 85 Supported notification types

Notification type	MWI switch setting	Mitel 415/430	Mitel SMBC	Mitel 470
Switched off	0			
Polarity reversal	− and +	–	✓	✓
High voltage	HV	–	✓	–
Frequency reversal (FSK)	No symbol (Any switch setting)	✓	✓	✓

Tip for the setting polarity reversal:
Set the switch of the phone (e. g. Mitel 6730 Analogue) to the symbol "−". If the MWI LED is blinking when a message is available and off when no message is available, the

switch is set correctly. If the MWI LED is on when a message is available and blinking when no message is available the switch must be set to “+”.

Notes:

- For the notification type FSK, a new message with a small envelope is displayed on the screen of the phone Mitel 6730 Analogue. This variant is not recommended as the symbol can be easily overlooked.
- The information in this section basically applies to analogue phones Aastra 1910 and Aastra 1930 too. In these models the MWI switch is labelled on the rear side of the phone, and the switch settings for polarity reversal, with PR1 and PR2.
- The notification type *Low voltage* is also supported (used for other analogue phones, especially in USA and Canada).

Mounting the phone on the desktop

Feed the mounting feet into the corresponding cut-outs on the underside of the phone until they engage. 4 different set-up angles are possible, by choosing the cut-outs and turning the set-up feet.

Mounting the phone on the wall

1. Place the supplied drilling template for wall mounting on the wall position you want and mark the positions for the mounting screws. Depending on the type of wall, you may need some dowel plugs. Screws and dowel plugs are part of the delivery.
2. Put the telephone with the mounting openings over the heads of the wall screws and pull the phone downwards to stop it.
3. On the cradle is a small clamp which is flush with the cradle surface. Push it up with a small flat head screwdriver and remove it from the phone.
4. With the cleat arm towards you and the flat side of the clamp towards the phone turn the clamp 180° and push it again into the cut-out in the phone cradle. Press in the clamp till it is flush with the surface and only the feet of the clamp are protruding.

Configuring keys

Configure the keys on analogue phones Mitel 6700 Analogue in the WebAdmin terminal configuration. The phone must be connected during configuration so the key configuration can be stored on the phone immediately. If not, you can load the key configuration on the phone after connecting the phone, by clicking *Update key configuration on phone*.

To load the key configuration on all connected Mitel 6700 Analogue series phones, click *Update key configuration for all Mitel 6700 Analogue phones*.

To load the key configuration stored in the WebAdmin from the connected phone, dial the function code *#53.

Labelling the phone

1. Remove the cover with the logo on top of the control panel by pressing down slightly and pushing up.
2. Pull out the designation label on the lugs, label it then push it back again into the cut-out.
3. Carefully put back the cover with the logo, so that the paper lugs are covered.

Powering the phone

The phone is powered via the FXS line.

5 Configuration

This chapter describes the web-based configuration tool WebAdmin as well as some additional options.

With WebAdmin the installer configures and maintains the MiVoice Office 400 communication server and its auxiliary equipment, and is supported in the process by a set up and configuration assistant. WebAdmin offers different user interfaces for administrators, system assistants and end-users as well as a special application for accommodation and hotels. A context-sensitive online help provides valuable instructions on configuration, and step-by-step instructions.

The chapter ends with valuable information and instructions on how to configure your MiVoice Office 400 communication system.

5. 1 WebAdmin Configuration Tool

This web-based configuration tool is available for the online configuration of MiVoice Office 400 series communication servers. It offers a simple, user-friendly interface and an online help, and with its different authorization levels it is aimed at different user groups:

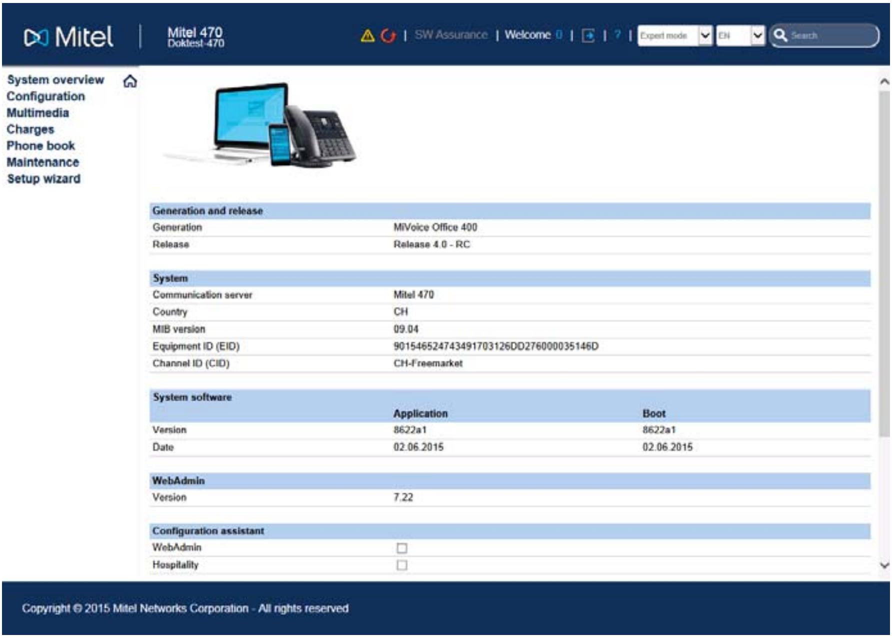


Fig. 73 WebAdmin Configuration Tool

Administrator authorization level:
The Administrator has access to all the views and functions of the configuration tool (*Expert mode*). He can call up a set-up assistant, show a general configuration assistant and a special hospitality configuration assistant, and configure all system parameters. The administrator can switch back and forth between *Expert mode* and *Standard mode* at any time.

Authorization level *Administrator (Standard mode only)*:
In Standard mode the administrator has access to all the main views and functions of the configuration tool. He can call up a set-up assistant, show a general configuration assistant and configure the most needed system parameters.

System assistant authorization level:
The System Assistant only sees selected views of the configuration tool, and the scope of functions is limited.

Hospitality-Administrator authorization level:
The Hospitality Administrator features all the views required to set up the Mitel 400 Hospitality Manager and the reception menu of the Mitel 6940 SIP, Mitel 6873 SIP or MiVoice 5380 / 5380 IP and specify its default settings. A link can also be used to start the Mitel 400 Hospitality Manager (see "Mitel 400 Hospitality Manager", page 178).

Receptionist authorization level:
This access starts the Mitel 400 Hospitality Manager directly (see "Mitel 400 Hospitality Manager", page 178).

The WebAdmin is included in the file system of each communication server of the MiVoice Office 400 family and does not have to be installed separately.

Access:
To access the login page of WebAdmin, enter the communication server IP address in your browser. You can find the registration data of a new communication server in the chapter "Default user account for initial access", page 183.

If you do not know the communication server IP address, search for the communication server on the IP network with the auxiliary application System Search (see page 180).

Note:
With the web-based administration two users are able to access the same communication server simultaneously (and no fewer than five users at the Receptionist authorization level). This can cause confusion if a configuration is being carried out in the same places.

5. 1. 1 Integrated and auxiliary applications

Mitel 400 Hospitality Manager

The Mitel 400 Hospitality Manager is a web-based application for receptionists in the hospitality sector. It provides a clear, at-a-glance list view or floor-by-floor view of the rooms and features functions such as check-in, check-out, notification, wake-up call, retrieval of call charges, maintenance list, etc.

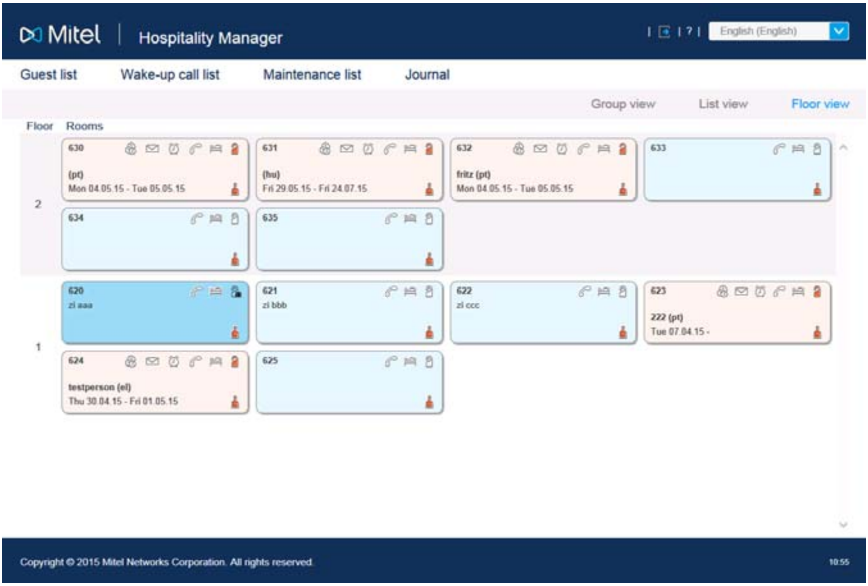


Fig. 74 Mitel 400 Hospitality Manager

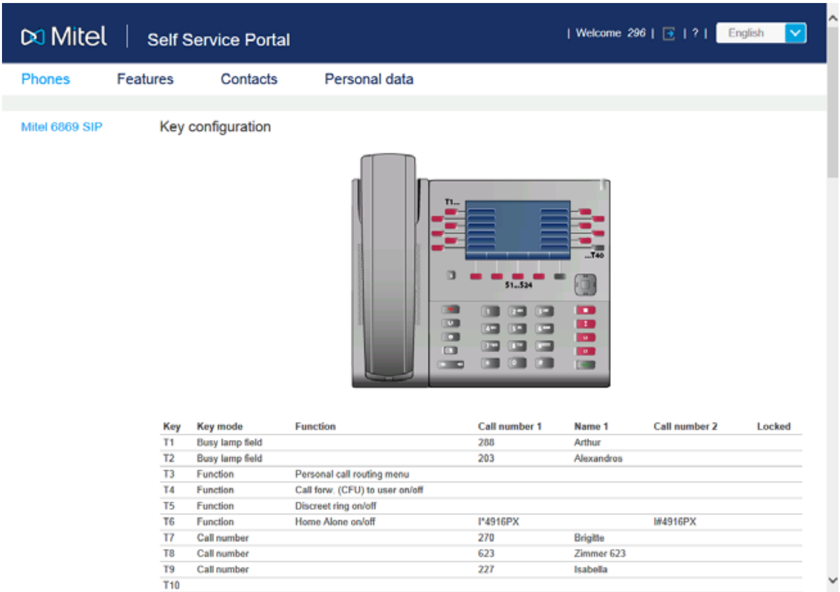
Mitel 400 Hospitality Manager is integrated into WebAdmin and subject to a licence.

Access:
You have access to two types in Mitel 400 Hospitality Manager:

- Register on the WebAdmin registration page with the access data of a user account to which an authorisation profile with the WebAdmin authorisation level *Receptionist* has been assigned. This starts Mitel 400 Hospitality Manager directly.
- Register on the WebAdmin registration page with the access data of a user account to which an authorisation profile with the WebAdmin authorisation level *Hospitality administrator* has been assigned. Click the menu tree on the left side on the *Hospitality Manager* input.

Self Service Portal

With the Self Service Portal, users can configure and adjust personal phone settings, such as key configuration, labels, display language, directly and independently on the PC. Users also have access to their personal mail boxes; they can configure and control presence profiles, personal call routing and call transfers, and create or search for private phone book contacts.



Key	Key mode	Function	Call number 1	Name 1	Call number 2	Locked
T1	Busy lamp field		288	Arthur		
T2	Busy lamp field		203	Alexandros		
T3	Function	Personal call routing menu				
T4	Function	Call forw. (CFU) to user on/off				
T5	Function	Discreet ring on/off				
T6	Function	Home Alone on/off	!*4916PX		!#4916PX	
T7	Call number		270	Brigitte		
T8	Call number		623	Zimmer 623		
T9	Call number		227	Isabella		
T10						

Fig. 75 Self Service Portal

The Self Service Portal application is integrated into WebAdmin.

Access:
You can access a user's Self Service Portal by entering any of the following combinations (registration data) on the WebAdmin registration page:

- Call number + PIN
- Windows user name + PIN
- Windows user name + password

The standard PIN "0000" is accepted, but must be changed during first login. You can choose any 2 to 10-digit number combination.

System Search

The auxiliary application System Search is an independent help tool for detecting MiVoice Office 400 series communication servers on the IP network. System Search MiVoice Office 400 finds all communication servers connected to the IP network, provided they are located on the same subnet as the PC and are at least compatible with Software release 1.0. (does not apply to Virtual Appliance). With System Search you can also see the name, type, sales channel, EID number and operating mode of a selected communication server. You can modify its IP address or directly start the WebAdmin administration tool.

Moreover, with System Search you can load language files for the audio guide, Mitel phones as well as for the user interface and online help of WebAdmin, Hospitality Manager and Self Service Portal via MiVoice Office 400 FTP server onto your PC and upload them afterwards to the communication server with WebAdmin. Thus, an update or an upload of new languages is possible without an internet connection of the communication server.

With System Search you can also upload system software in boot mode (Emergency Upload). This is particularly useful if the current software application on the communication server is no longer able to run or if you wish to load an older software application (does not apply to Virtual Appliance).

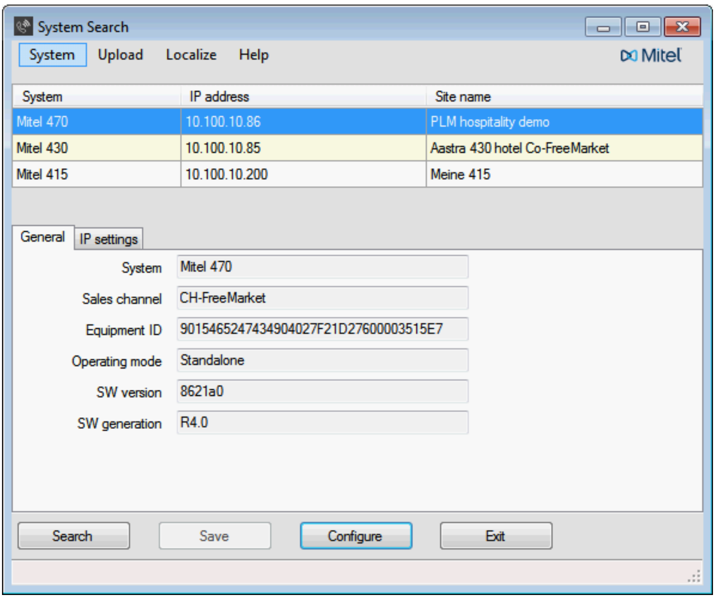


Fig. 76 System Search

You can download the System Search application via Software download server. For this, you must first log on to the Extranet with your partner login. The application must not be installed but is started with a double-click.

Note:
For Virtual Appliance and SMB Controller, System Search is only available for downloading language files for the audio guide, Mitel SIP terminals as well as for the WebAdmin, Hospitality Manager and Self Service Portal user interfaces and online help.

Mitel 400 WAV Converter

The auxiliary application Mitel 400 WAV Converter is an independent help tool for compressing audio data. If the integrated voice mail system is operated in expanded mode (Mitel 415/430 only), all the audio data must be available in compressed G.729 format. To be able to continue using existing, uncompressed greetings in G.711 format, you need to compress them first. Mitel 400 WAV Converter is available for this.

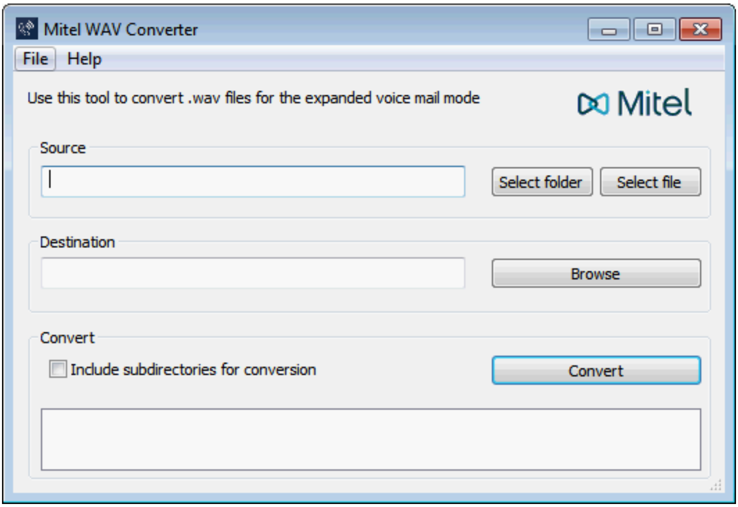


Fig. 77 Mitel 400 WAV Converter

You can download the application via Software download server. For this, you must first log on to the Extranet with your partner login.

The application must not be installed but is started with a double-click.

5. 2 Access types with WebAdmin

There are the following possibilities to access the MiVoice Office 400 communication server with WebAdmin:

- In the LAN with an Ethernet cable (directly or via a switch)
- Externally via SRM (secure IP remote management)

Note:
External access (ISDN/analogue) with a dial-up connection is only recommended on some conditions, for performance reasons.

First access on LAN

For a first access to the communication server, it is easier if your computer is located on the same subnet as the PC. If this is not the case, you can also connect the computer directly to the communication server using a LAN cable.

With the auxiliary application System Search (see page 180) the communication server (and other MiVoice Office 400 series communication servers on the same subnet) is searched and displayed. It is advisable to directly deactivate the normally activated communication server DHCP via System Search and to manually enter a static IP address, the subnet mask and IP gateway. After login via the standard access (see "Default user account for initial access", page 183), the data is stored in the communication server.

See also:
If you are setting up an MiVoice Office 400 communication system for the first time, read the chapter "Getting started", page 36).

Accessing the communication server on the LAN

If the communication server IP address is known, it can be entered directly in the address line of a web browser. WebAdmin is started after the access data is entered. The computer only needs to be located on the same LAN, but not necessarily on the same subnet.

Accessing the communication server from outside

For remote access to the communication server, we recommend SRM (Secure IP Remote Management) secure IP remote management. For this, you need to install an SRM agent on your computer with which you can set up a connection to the SRM server. Thereafter, the SRM server calls the communication server via PSTN and sends it the connection parameters. The communication server now sets up a secure connection to the SRM server which switches together them with the connection to the SRM agent.

See also:
You can find instructions on how to set up Secure IP Remote Management in the WebAdmin help on the *IP remote management (SRM)* view (*=mw*).

5. 3 User access control

Access to the configuration is password-protected. Any user wanting to log in to a communication server is prompted for his user name and password (access data).

5. 3. 1 WebAdmin User accounts and authorization profiles

A user's authorizations are regulated by authorization profiles, which are assigned to the user accounts.

5. 3. 1. 1 User accounts

Default user account for initial access

When a new communication server is opened or after a first start, the default user account (*admin*) and several authorization profiles are created. The default user account is linked with the authorization profile *Administrator*. This authorization profile is assigned the administration rights for the *User access control* for *Audio services* and for WebAdmin at the *Administrator* authorisation level.

The required user accounts and authorization profiles can be set up using the default user account.

To access the default user account (*Default User Account*) enter the following:

Tab. 86 Standard user account and standard password

User name	admin
Password	password

Other predefined user accounts

The predefined default user account *SystemUserInterface* is used to control access via the control panel for the colour display on the front panel. Access is PIN-protected (see "Call-Manager display and control panel", page 218).

The predefined user account *amcc* is meant for operating a Mitel Mobile Client Controller.

The two predefined user accounts *blustar* and *bucs* are meant for BluStar terminals and for a BluStar server.

Furthermore there are predefined user accounts for the Mitel Dialer for MiCollab and for OpenMobilityManager (OMM).

You can see the predefined user accounts in the *User account* (*=a7*) view.

Note:
The predefined user accounts cannot be deleted.

Personal user accounts

Subject to the administration right for user access control, personal user accounts can be created in user access control (*=a7*) and assigned some authorisation profiles. The following rules apply to user names selection and spelling:

- An user name must consist of a minimum of 1 and a maximum of 25 alphanumerical characters.
- Unlike the passwords, the user names are **not** case sensitive.
- The following special characters can be used: ?, /, <, >, -, +, *, #, =, full stop, comma and space.
- German umlauts (e.g. ä, ö, ü) and other diacritical characters (e.g. é, à, â) are not permitted.
- User names must be unique throughout the system.
- The user name and password must not be identical.

5. 3. 1. 2 Authorization profiles

Predefined authorization profiles

The predefined authorization profiles are assigned administration rights and interface user rights. An overview of all predefined authorisation profiles with their administration and access rights is available in the WebAdmin help on the *Authorization profile* view(*=u5*).

Personal authorisation profiles

Subject to administration right for the administration right for user access control, no personal authorisation profiles can be protected and assigned the desired rights. A description of the various administration and access rights is available in the WebAdmin help on the *Authorisation profile* view(*=u5*).

Note:
Authorization profiles can only be viewed or created by *Administrators* in *Expert mode*.

5. 3. 1. 3 Passwords

To ensure that the communication server can only be configured by authorized personnel, access to the configuration is password-protected.

Password syntax

The following rules apply to password selection and spelling:

- A password must consist of a minimum of 8 and a maximum of 255 characters.
- Unlike the user names, the passwords are case sensitive.
- The passwort must contain at least one uppercase letter A - Z.
- The passwort must contain at least one lowercase letter a - z.
- The passwort must contain at least one digit 0 - 9.
- The passwort must contain at least one of the following special characters: ?, /, <, >, -, +, *, #, =, full stop, comma and space.
- German umlauts (e.g. ä, ö, ü) and other diacritical characters (e.g. é, à, â) are not permitted.
- The default password *password* is not permitted.
- The password must not be the same as the user name.
- It is not allowed to use the last 4 historic passwords.

Change password

Any user who has been assigned an authorisation profile in which the *User access control* administration right is released is authorised to modify the passwords of all user accounts. It is therefore advisable to assign this administration right restrictively.

Users whose password has been changed are prompted to enter their newly assigned password the next time they log in. The same applies to users whose accounts have been newly created.

Users without the administration right *User access control* can only change their own password.

Access with incorrect password

After 15 failed login attempts using incorrect passwords the corresponding user account is blocked; it can then only be reactivated by a user with the *User access control* administration right. He then replaces the old password with a new one. The next time he logs in, the corresponding user is prompted to change the password and enter the new one he has been assigned.

Lost password

If another user has also been defined with the *User access control* administration right released, he can simply overwrite with a new password the password lost by another user. The next time he logs in, the corresponding user is prompted to change the password and enter the new one he has been assigned.

If the passwords of all administrators are lost, access can still be gained locally without a password (see "Password-free access", page 186).

5. 3. 2 Password-free access

It is possible to activate on the front panel a function that enables via password-free, local access via LAN with administration right *User access control*. This is useful for example if all the passwords have been lost.

There is no password-free access for remote maintenance.

5. 3. 3 Automatic exit from the configuration

Access to the configuration is interrupted if no changes are made to a parameter value or the navigation system is not used during a specific timeout.

5. 3. 4 WebAdmin access log

An access log with 20 entries is drawn up for each user account so that the history of accesses to the configuration can be tracked. Denied access attempts using erroneous or incorrectly type passwords are also logged. The logs can be read by each user (authorization level) *Administrator* in *Expert mode* required).

Retrieving the log data

The system monitors all the accesses and failed access attempts and saves them in the file system of the communication server. These lists can be retrieved locally or remotely. (*=ez* or *=z3*).

CLIP verification

If in the general maintenance settings (*=t0*) of the parameter *CLIP required* is activated, remote maintenance is only possible if the retrieving party is using a CLIP. The CLIP number is also recorded by the access log.

Entering the processes in the log

Each access attempt generates an entry in the corresponding list.
In case of remote maintenance an entry will not be generated if remote maintenance is barred or if *CLIP required* is activated in the configuration and no CLIP is received.

5. 4 WebAdmin remote access

With a remote maintenance access the user is authenticated using his user name and password. The user account must also be assigned an authorization profile in which the interface access *Remote maintenance dial-up access* is enabled. This also applies to SRM (Secure IP Remote Management), secure IP remote management.

5. 4. 1 Access enabled by local users

Remote maintenance access can be enabled in two ways:

- Using function codes (see page 187)
- With WebAdmin

It can be revoked again automatically or manually.

All enabling types have equal authorization status. This means that remote maintenance access can be enabled using a function code for example, and then barred again using the WebAdmin in general maintenance settings (*=t0*).

When remote maintenance access is activated, the event message *Remote maintenance on* is sent to all message destinations where the corresponding filter criteria in the assigned event table is set accordingly (see chapter "Event tables", page 243).

If remote maintenance is released, this can be recognised in the WebAdmin title bar of the symbol.

Remote maintenance access can be enabled or barred using the function codes both from the idle state and the talk state, e.g. after an enquiry.

The authorisation to activate or bar remote maintenance access using the function code is defined and granted to the user with the parameter *Remote maintenance access* in a permission set (*=cb*).

After a first start of the communication server, the authorizations of all users are restricted.

Note:
It is advisable not to keep the remote maintenance access permanently activated. This ensures that the communication server data cannot be manipulated from a remote location by unauthorized persons.

5. 4. 2 Function code for remote maintenance access

Tab. 87 Function code for remote maintenance access

Enable/bar a one-off remote maintenance access	*754 / #754
Enable/bar a one-offpermanent maintenance access	*753 / #753

When remote maintenance access is enabled using function code *754, access will automatically be barred again once the remote maintenance process has been completed. It is possible to bar remote maintenance manually using #754 before it is initiated.

Remote maintenance access can be enabled permanently using the function code *753. To bar access, the authorized user must enter the function code #753 manually.

The enabling or barring of remote maintenance access using the function code is signalled in each case by an acknowledgement tone.

Remote maintenance access can also be enabled or barred in WebAdmin, if the relevant authorization has been given.

Note:
In a QSIG network it is important to ensure that the authorization to change the remote maintenance access is also denied to unauthorized PISN users. Otherwise, a PISN user would be able to use an abbreviated dialling number defined for the destination PINX and containing the appropriate function code to change the remote maintenance access to the destination PINX.

Mitel Advanced Intelligent Network:
In an AIN the remote maintenance access of all the nodes depends on the setting in the Master. If remote maintenance access is enabled in the Master, both the AIN configuration and the offline configuration of the satellites are enabled.

Remote maintenance access via an external dial-up connection to the AIN is also protected and has to be explicitly enabled via the control panel on the front panel. This is irrespective of whether dial-up access is via a satellite or directly to the Master.

5. 4. 3 Function keys for remote maintenance access

On system phones the function code for enabling/barring remote maintenance access can be stored under a function key, provided the user has the appropriate authorization.

The relevant LED lights up if remote maintenance access is enabled once or permanently.

The relevant LED goes off as soon as remote maintenance access is denied again, either automatically or manually, using the function code or WebAdmin.

5. 5 Configuring with WebAdmin

The configuration steps are based on the information determined during the planning and, where applicable, the installation.

Whenever possible, use the planning and ordering software Mitel CPQ, to set up your communication system. Mitel CPQ can be operated online after logging in at Mitel Connect https://connect.mitel.com. Mitel CPQ not only calculates the required hardware – it also lists the required licences for the planned operation.

See also:
If you are setting up an MiVoice Office 400 communication system for the first time, read the chapter "Getting started", page 36.

Setup wizard

The WebAdmin setup wizard takes you step by step through the setup of a basic configuration and is suitable for initial communication server setup. The setup wizard is automatically called up when a new communication server is installed. Logging on as administrator in WebAdmin (expert or default mode) allows you to also start the setup wizard directly from the WebAdmin navigation tree.

The setup wizard comprises the following steps:

1. Activating licences
2. Setting up the IP addressing
3. Configuring media resources
4. Setting up the numbering plan
5. Setting up SIP providers
6. Setting up users, terminals and DDIs
7. Setting up the auto attendant

For each step you can display a help page or see it in the lower part of the window where it is already displayed. You can skip individual steps of the setup wizard or exit the setup wizard at any time in order to return to the WebAdmin start page.

Configuration assistant

The configuration assistant goes further than the setup wizard and helps you to configure a communication system in sequence, from scratch. Logging on as administrator in WebAdmin (expert or default mode) allows you to display the configuration assistant on the WebAdmin start page.

The configuration assistant comprises the following steps:

1. Setting up the IP addressing
2. Regulating access control
3. Checking licences
4. Configuring media resources
5. Setting time and date
6. Checking network interfaces
7. Setting up SIP providers and accounts
8. Specifying user permissions

9. Create users and DDI[1)] numbers
10. Checking outgoing routing
11. Setting up the auto attendant
12. Setting up music on hold
13. Setting up an announcement service
14. Entering abbreviated dialling contacts
15. Saving configuration data

For each step, the upper half of the screen displays the configuration overview; the right-hand side contains notes and instructions about the step you have selected. The WebAdmin online help can be called up for further help.

You can skip individual configuration assistant steps or call up additional views of the WebAdmin navigation tree. To hide the configuration assistant again, untick the control box on the WebAdmin start page.

Configuring the CPU2-S application card

The configuration of the application card is described in detail in the Installation Instructions for CPU2-S application card.

5. 6 WebAdmin Configuration Notes

The sections below contain information that may be useful before, during or after a configuration with WebAdmin.

5. 6. 1 Licences

All the features (even those subject to licences) can be configured without a valid licence.

If you use a function or feature that requires a licence but do not actually have the relevant licence, a trial licence is acquired automatically; it is also shown in the overview of activated licences (*Licences* *=q9* view). With a trial licence you can now use the function or feature free of charge for 60 days. The trial licence's expiry date is indicated under *Status*. This procedure can only be used once for each function or feature. Thereafter you must acquire a licence. The licence overview (Tab. 33) shows which trial licences are available.

All licences are stored in a licence file, which you can obtain from your authorised dealer. Each licence file can only be used for one communication server. To licence several communication servers, you will obtain separate licence files to match the licence information of the individual communication server. If a communication system

1) In USA/Canada the abbreviation DID (Direct Inward Dial) is used instead of DDI (Direct Dialling In).

consists of several communication servers (e.g. in a AIN), normally only one licence file is required on the Master.

A new communication system must be activated first after commissioning. Otherwise, the communication server changes after 4 operating hours to limited operating mode.

Upload the licence file in the *Licences* (=*q9*) view.

If you have received a voucher (or with the help of the *Equipment ID*), you can also obtain the licence file via Mitel Connect https://connect.mitel.com (partner login required). You can find instructions about this in WebAdmin help.

See also:
"Licences", page 74

5. 6. 2 File management

The file management of the MiVoice Office 400 application is done via WebAdmin:

- *Localization* (=*e6*)
  You can adapt the communication system to your country's specifications, with the help of localization. In this view language files can be manually or automatically loaded for Mitel 6800/6900 SIP phones via FTP server. Moreover, you can manually or automatically load the languages for the WebAdmin, Hospitality Manager and Self Service Portal user interface and online help, as well as an external numbering plan for the SIP connection via the FTP server.
- *File system state* (=*e3*)
  In this view you can see the thematically structured file system's memory load. In an AIN the file systems for all nodes can be viewed.
- *File browser* (=*2s*)
  With the file browser you have access to the communication server file system and create new folders as well as view, import, replace or delete files in the file system.

Note:
File management is only accessible for *Administrators* in *Expert mode*.

See also:
You can find detailed information about the functions in WebAdmin help for the corresponding view.

5. 6. 3 System reset

5. 6. 3. 1 Restart

Restart via WebAdmin

A restart via WebAdmin is triggered in the maintenance settings with the *Restart* button in the *System reset*(=4e) view.

A restart via WebAdmin reboots the MiVoice Office 400 communication server. The configuration data is preserved.

Restart via front panel

A restart via front panel is done using the control panel. The configuration data is preserved (see "Call-Manager display and control panel", page 218).

Notes:

– Never disconnect the communication server from the power supply to trigger a restart. This can result in data losses and prevent a restart.
– The restart is triggered immediately. All the active call and data connections are interrupted.

5. 6. 3. 2 First start

A first start has the effect of resetting the MiVoice Office 400 communication server from scratch. The system-specific data such as the system ID, system type, sales channel, licence file, software generation and IP address of the system are preserved.

Notes:

– A first start deletes all the configuration data already stored and replaces it with the default values of the sales channel. Therefore, back up your configuration data before a first start.
– The first start is triggered immediately. All the active call and data connections are interrupted.

First start via WebAdmin

A first start via WebAdmin is triggered in the maintenance settings with the *First start* button in the *System reset*(=4e) view.

First start via front panel

A first start via front panel is done using the control panel (see See "Call-Manager display and control panel", page 218).

First start and reset sales channel via WebAdminl

With the *First start and reset sales channel* button in the maintenance settings of the WebAdmin *System reset* (=4e) view, you have the possibility not only to execute a first start but also to delete the sales channel. During the next start, you will be prompted for the sales channel and licence file. Note that the licence file is dependent on the sales channel. This means you can no longer use the existing licence file, if you choose another sales channel.

Note:
This function is only accessible for Administrators in Expert mode.

5. 6. 4 Data backup

With a configuration data backup all the MiVoice Office 400 configuration data of the communication server is stored in a compressed file in ZIP format. You can let the configuration data backup run automatically (*Auto backup*) or as required (*Manual backup*).

You can automatically copy the backup files to an FTP server or e-mail them.

With an audio data backup all the audio data of the communication server is backed up in a compressed file in ZIP format. The backup of the audio data can only be done manually.

You can find the automatic data backup and distribution service settings in the WebAdmin *Maintenance* / *Data backup* (*=um*) view where you can also test the configuration. Moreover, in this view, you can see the automatically and manually created backup files and also restore or delete them.

The configuration backup and the audio data backup are always stored in a encrypted format.

Note:
The backup may consist of several files. They are compiled by the communication server and compressed into a ZIP file. During the Restore process the ZIP file is extracted by the communication server itself. To ensure the restore process to run smoothly, make sure you do not modify the ZIP file. Never extract or modify a backup file yourself.

5. 6. 4. 1 Auto backup

The automatic data backup function creates a backup of the MiVoice Office 400 configuration data at regular intervals and saves the backup files on the communication server's file management system.

The Auto Backup function creates a backup of the configuration data at daily, weekly and monthly intervals:

- Everyday at the set time a backup is created and stored in the *..\backup\day* directory.

- When the week changes, a copy of the backup is stored in the *..\backup\week* directory.
- When the month changes, a copy of the backup is stored in the *..\backup\month* directory.

The backup directories are located on the file system of the communication server and are directly accessible via the *File browser* (*=2s*) or with an FTP connection.

A backup remains stored until the set storage time has expired; the .zip file is then deleted from the file system.

5. 6. 4. 2 Distribution service

You can use the distribution service to automatically copy the backup files to an FTP server or e-mail them.

- The e-mail distribution service sends a copy of each backup file created to a preconfigured e-mail address.
- The FTP distribution service stores a copy of each backup file created on an FTP server.

5. 6. 4. 3 Manual backup

Configuration and audio data must be stored separately and stored as .zip files on any data carrier you want. The configuration data is also backed up automatically as copy on the communication server file system.

Situations in which you have to create a manual backup:

- Before running a first-start of the communication server (a first start resets all the configuration data to their default values and deletes all audio data).
- Before and after you have expanded (or reduced) the communication server with cards or modules.
- Before and after any major configuration changes.

5. 6. 4. 4 Restore backup

The available MiVoice Office 400 configuration data and audio data backup files can be restored at any time.

Note:

– Restoring a backup irretrievably overwrites the current configuration data or audio data.

– Restoring a backup also resets the users' presence status, the personal routing settings and any activated CFUs to the backup status.

– Some configuration changes only take effect after a restart. The communication server is restarted after the configuration data is restored.

See also:
The procedure for creating and restoring a backup is described in detail in the WebAdmin help in the *Data backup* (*=um*) view.

5. 6. 5 Importing and exporting configuration data

You have the possibility to edit various configuration data outside WebAdmin, or to import configuration data from other MiVoice Office 400 series communication systems. Here you can create, with the help of the export function, a specific Excel file hereinafter referred to as *Export file*. The export file contains several spreadsheets. Each sheet covers a specific configuration area. Subsequently, edit then re-import the export file. Only the data belonging to the view, on which you have activated the import function, will be imported. Example: The import function in the *Phone book* / *Public* view imports only the data from the export file located on the spreadsheet *Abbreviated dialling list*.

Exception: The export function in the *Backup* view imports the data in all spreadsheets.

You can find the export function in the following views:

- *Overview* (user data and key configuration of the terminals)
- *Abbreviated dialling numbers*
- *PISN user*
- *Time controlled functions*
- *Ext./Int. Allocation*
- *LCR*
- *Blacklist*
- *CLIP based routing*
- *Data backup*

Note:
You can activate the *Replace existing configuration* option with the import function. Activate this function only if you are setting up the communication server from scratch. This action deletes all previously configured user data and all user associated settings such as DDI numbers, CDE targets, user group entries, assigned phones, configured keys, etc.

5. 6. 6 Mitel 6800/6900 SIP phones

Prior to the registration, reset any phones that were already in operation back to the factory setting. For security reasons, delete the phone's MAC address in WebAdmin. This prevents problems during registration.

Use these procedures in the following cases:

- Assigning the phone to another user on the same system
- Transferring the phone to another system with the same software release
- Changing the software release to an earlier release
- Changing the communication server IP address

6 Operation and Maintenance

This chapter describes maintaining the system and configuration data as well as updating the system software. Replacing cards, modules and terminals are also described. The display and control panel of the communication server as well as operations supervision using the event message concept, the operating state display, and the error display are also topics covered in this chapter.

6. 1 Data Maintenance

6. 1. 1 What data is stored where

The communication server's data storage system consists of different elements:

- In the Flash components are stored the system software, the boot software and the configuration data. The contents of the memory are retained even when there is no power supply.
- In the RAM components (main memory) are stored volatile data that cannot be saved. It is available only when the system is in operation.
- The EIM card (Equipment Identification Module) contains the system-specific data (system ID, system type, sales channel, generation, DECT identification numbers, IP address of the configuration server). The contents of the memory are retained even when there is no power supply.
- The data of applications on the applications server (if a CPU2-S applications card is fitted) is stored on a hard disk.

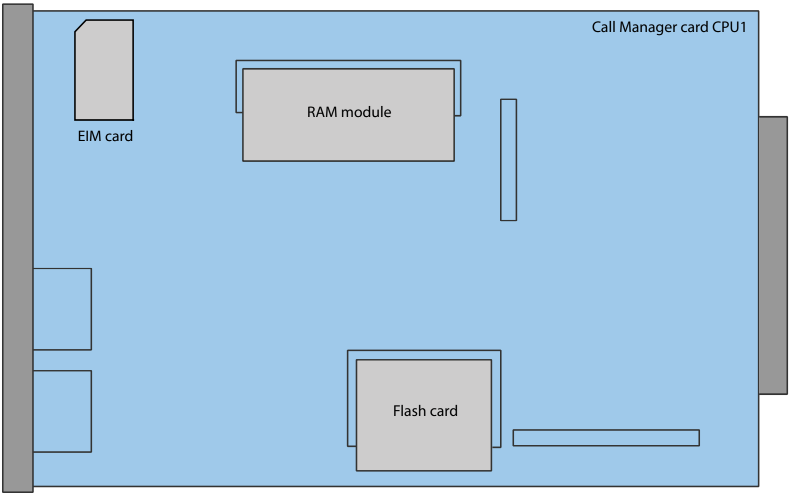


Fig. 78 Memory of the call manager card CPU1

6. 1. 1. 1 System software

The communication server's entire system software package is stored in compressed form in the Flash memory.

The RAM components comprise the main memory for program data. When the communication server starts up, the software on the Flash memory is decompressed, loaded into the main memory and started.

6. 1. 1. 2 File system

MiVoice Office 400 file system

The file system of the MiVoice Office 400 communication server comprises the MiVoice Office 400 application software, the software for system phones, the system and terminal configuration data, the audio data, system logs, data for WebAdmin, etc. With WebAdmin you have access to the file system via the menu item *File management*. You can see the file system memory load and you can load audio data, languages for the user interface and online help, language files for Mitel 6800/6900 SIP-series phones as well as an external numbering plan for SIP connection. Moreover, with the file browser you have the possibility to view, upload, replace or delete folders and files in the file system. (see also "File management", page 191).

Functions for backing up and restoring MiVoice Office 400 configuration data and audio data are available in the WebAdmin*Maintenance* / *Data backup* (*=um*) view (see also "Data backup", page 193).

Usually there is no need to access the MiVoice Office 400 file system directly as all needed functions are available in WebAdmin. For special cases you can access the MiVoice Office 400 file system with a FTP session.

Note:
Modifying or deleting files on the file system can result in a system that is no longer able to run.

6. 1. 1. 3 Boot software

The boot software is stored in a different Flash memory, which allows the communication server to start up in the boot mode, even if without executable MiVoice Office 400 application software.

6. 1. 1. 4 System-specific data

The system-specific data (system ID, system type, sales channel, generation, DECT identification numbers, IP address of the communication server) is stored on the EIM card (chip card). This data is not deleted by a first start of the communication server, and remains available. It can be ported to a different communication server by replacing the EIM card.

6. 1. 2 Updating configuration data

There are system-wide, user-related and terminal-related configuration data:

- System-wide configuration data can only be modified with WebAdmin.
- Terminal configuration data such as key assignments or ringing melodies can be modified either directly on the terminal, with Self Service Portal or with WebAdmin. With some system phones configuration is also possible using the web user interface or with the help of configuration files.
- User-related configuration data such as private contacts or CFUs is valid for all the terminals assigned to the user and can be configured using WebAdmin, partly via Self Service Portal, or directly on the terminal itself.

Access to the configuration data via WebAdmin is regulated by a User Access Control with user accounts, authorization profiles and authorization levels. More information can be found in the Chapter "User access control", page 183.

6. 2 Update Software

6. 2. 1 System software

MiVoice Office 400 application software

The MiVoice Office 400 application software is normally updated with WebAdmin. In exceptional cases (e.g. during downgrade), an Emergency Upload via System Search is required (see also page 201).

Firmware for system terminals

The firmware for MiVoice 5300/MiVoice 5300 IP, Mitel 600 DECT phones, DECT phone Office 135/135pro, DECT radio units SB-4+/SB-8/SB-8ANT and WebAdmin is also available in the MiVoice Office 400 application software.

Tip

The communication server software version can be displayed as follows on MiVoice 5300/MiVoice 5300 IP phones:

1. Access the configuration menu *Settings*.
2. Long-click on the * key

Information can be retrieved on Mitel 6800/6900 SIP phones as well as on Mitel 600 DECT DECT phones via the menu.

Depending on the phone, additional information is displayed.

Providing the MiVoice Office 400 system software and licence file

The new MiVoice Office 400 system software and the relevant licence file are provided by your sales dealer. In most cases you will download the software from an internet site specified by your sales partner. You will also receive a voucher. With this you can generate the new licence file through the Mitel Connect internet portal https://connect.mitel.com and upload it to your communication system. You need a login to access Mitel Connect (user name and password).

Load new MiVoice Office 400 system software with WebAdmin

New MiVoice Office 400 system software can conveniently and safely be loaded on the communication server file system in the WebAdmin *Maintenance* / *System software* (*=m7*) view. The activation point of the new software is selectable. (Exception: The activation time on the satellites AIN always depends on the master's demand).

In newly delivered systems it is possible to directly load new system software after choosing the sales channel.

Notes:

- Most times a new licence file is also required for new MiVoice Office 400 system software. You can also install and start up the new software without specifying the licence file. However, once you have started to use the software you will need to upload the licence file within 4 hours; otherwise the communication server will switch over to the restricted operating mode. In this mode, only the basic functions of the communication server are available.
- Depending on communication server type, the upload operation (especially decompressing the software package) may take some time.
- Never disconnect the communication server from the power supply during the update process. This may prevent executable system software from being available on the communication server, and make an EUL (Emergency Upload) necessary.
- Please read the chapter “Important hints and restrictions” in the release notes to the software to be loaded.

See also:
A detailed description of the software upload procedure with WebAdmin is available in the online help.

Loading new or older system software with System Search

Whenever a standard software upload is not possible, has proved faulty, or to replace a Flash card or if you wish to load an earlier system software (Downgrade), you must carry out a Emergency Upload. You need the search and help tool System Search.

Note:
A first start of the communication server is also performed with an Emergency Upload. All the already stored configuration data is deleted and replaced with the default values of the sales channel. Therefore, before an Emergency Upload back up your configuration data (if still possible).

To perform an Emergency Upload, proceed as follows:

1. Set the communication server to boot mode using the navigation key (see "Boot mode", page 220).
2. Start System Search and select *Emergency Upload*.
3. Enter the communication server IP address.
4. Select the system software package to be uploaded (zip file).
5. Click the *Upload* button.
   → Emergency Upload is started.

6. 2. 2 Firmware for corded system phones

The MiVoice Office 400 application software package contains the software for certain system phones (DSI and IP), which is updated in each case along with the application software. For other system phones (SIP) the firmware is located on a firmware server.

The MiVoice 5360 system phones do not have their own memory. All other system phones have a Flash memory.

SIP system phones

The firmware for Mitel 6800/6900 SIP phones as well as for Mitel BluStar 8000i, Mitel BluStar clients and Mitel Dialer is preferably located on a firmware server. In the WebAdmin *Configuration* / *IP network* / *Firmware server* (*=yv*) view Mitel FTP servers are already predefined. Various firmware releases are stored on this server, according to different communication server software releases. The predefined entry in WebAdmin is adjusted to each communication server release if necessary. You can also indicate the address of another firmware server.

Whenever the phones are started the phone firmware version is compared with the version on the firmware server. If the versions differ, the firmware is downloaded from the firmware server to the phones.

DSI and IP system phones with Flash memory

The flash memory contains the boot software and the application software. DSI phones also have an area with the interface software.

The firmware for the phones MiVoice 5370, MiVoice 5380 as well as for all MiVoice 5300 IP series phones is contained in the MiVoice Office 400 application software package. The firmware versions are compared when the phones are started. If the versions differ, the firmware is downloaded from the communication server to the phones. When updating the system software this can take several minutes for each DSI phone.

The expansion key modules MiVoice M530 and MiVoice M535 also have a flash chip containing firmware. The update mechanism is the same as the one described above. However a local power supply is always required (Power over Ethernet is also possible with IP terminals).

6. 2. 3 Firmware System MiVoice Office 400 DECT

DECT radio units SB-4+, SB-8 and SB-8ANT

The Flash memory on the radio units contains an area that cannot be modified. It is used for starting the radio unit and receiving the firmware for the radio unit.

The actual firmware for the radio unit is contained in the MiVoice Office 400 application software package. The loaded firmware is tested when the radio unit starts up. If the loaded firmware is not identical to the version in the system software, the firmware will be downloaded from the communication server on to the radio unit and stored in the Flash memory of the radio unit.

Cordless DECT phones of the Mitel 600 DECT family

The firmware for the Mitel 600 DECT cordless phones, is updated via radio (Air-Download). The update can be enabled or disabled individually for each cordless phone using the menu *System* - *Download server* on the cordless phones. If the cordless phone is logged on to several systems, this menu defines which system the firmware update is relevant to.

There is only one firmware for the cordless Mitel 600 DECT series phones. It is included in the MiVoice Office 400 application software package and stored in the file system of the communication server.

DECT cordless phones Office 135 and Office 160

The firmware for the Office 135 and Office 160 cordless phones, is updated via radio (Air-Download). This requires the cordless phone to be logged on to system A.

The memory in the cordless phones is a Flash memory. The Flash memory contains an area that cannot be modified. This area contains the cordless phone's boot software.

The firmware for the cordless phones is contained in the MiVoice Office 400 application software package. The loaded firmware is tested when the cordless phone starts up. If the loaded firmware is not identical to the version in the system software, the system will initiate an Air-Download. The firmware is loaded from the communication server onto the cordless phones via radio and stored in the Flash memory.

To be able to run an Air-Download, you need to ensure that the cordless phone contains a functional firmware.

The cordless phone remains fully functional during an Air-Download. The new loaded firmware is activated only once the Air-Download has been successfully completed. A restart is carried out on the cordless phone.

6. 2. 4 Firmware System Mitel SIP-DECT

With Mitel SIP-DECT and Mitel 600 DECT series phones comprehensive solutions can be provided for wireless telephony on IP-based networks. This requires RFP radio units that can be directly connected to other VoIP devices on the LAN. OpenMobilityManager (OMM) is installed on one of the RFP radio units or on a PC, which constitutes the management interface for the Mitel SIP-DECT solution. Mitel 600 DECT phones have loaded a different firmware in an Mitel SIP-DECT system from the one in an MiVoice Office 400 DECT system.

The firmware for the RFP radio units and for the Mitel 600 DECT cordless phones is preferably located on a firmware server. Automatic firmware update is then possible. The WebAdmin *Configuration* / *System* / *DECT/SIP-DECT* / *SIP-DECT* (*=9y*) view contains a global predefined Mitel FTP server. Various firmware versions are stored on this server, according to different communication server software releases. The predefined entry in WebAdmin is adjusted to each communication server release if necessary. You can also indicate the address of another firmware server.

Firmware designations for Mitel SIP-DECT (examples):

aafon6xxd.dnld:
Firmware for Mitel 600 DECT cordless DECT phones.

iprfp3G.dnld:
Firmware for OpenMobilityManager (OMM).

6. 2. 5 Applications card CPU2-S

The updating of the application card software is described in detail in the Installation Instructions for CPU2-S applications card.

6. 3 Hardware update

Hardware maintenance comprises replacing cards, modules and terminals when there is a defect or for a generation change. Safety regulations must be observed and the step-by-step procedure must be followed.

6. 3. 1 Preparations

The following preliminary steps apply to interface cards, system cards and system modules as well as to the call manager card of the communication server itself. The preliminary steps for replacing an applications card are described separately.

First steps before cards are removed or added:

1. Inform all concerned users if the system has to be put out of operation during working time.
2. Shut down the call manager via the control panel (see "On/Off key", page 218).

6. 3. 2 System information

Some system information are stored on the EIM (Equipment Identification Module) card. The information includes:

- The EID (Equipment Identification) serial number
- The sales channel identification CID (Channel Identification)
- The system type
- The application software generation
- The IP address of the MiVoice Office 400 communication server

The data is not deleted by a first start of the MiVoice Office 400 communication server, and remains available.

6. 3. 2. 1 Licences

To expand a system already in operation or to re-order a licence for a new system, proceed as follows:

1. Order the licences you want from your authorised dealer and specify the EID number, which serves to identify the communication server.
2. The new licence file can be obtained either from your authorized dealer or via Mitel Connect https://connect.mitel.com using the EID (partner login required).
3. Upload the licence file in the *Licences* (*=q9*) view. The licence file is stored in the file system of the communication server in the sub directory ...\data\lic.
4. The newly licensed features are enabled. It is not necessary to restart the communication server (exception: AIN licences).

See also:
"Licences", page 74

6. 3. 2. 2 EIM card

The EIM card must be replaced in the following cases:

- The call manager card is defective
- The EIM card is defective

The call manager card is defective

If a defective call manager card is replaced, the EIM card has to be switched from the defective call manager card to the new one. For instructions on how to replace the call manager card, see page 212.

The EIM card is defective

In the unlikely event of a defective EIM card, contact your authorized dealer to discuss the procedure.

For the procedure for switching an EIM card see page 210.

6. 3. 3 Interface cards

The different card types, the number of slots and the maximum configuration are all determined by the system capacity (see "3 Expansion Stages and System Capacity").

A number of rules have to be observed when fitting the cards (see "Component mounting rules", page 115.

All configuration data is centrally stored in non-volatile Flash memory. This means that configuration data is preserved whenever a defective interface card has to be replaced by a new one.

6. 3. 3. 1 Replacing a defective interface card

A card is replaced by the same card type with the same number of ports.

Procedure:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the defective interface card and remove the card by pulling the fastening screw.
3. Carefully slide the new interface card into the slot shaft and gently press the card as far as it goes into the connection on the backplane.
4. Use the screw to secure the card in its slot.
5. Restart the call manager by pressing the On/Off button on the call manager card.

6. 3. 3. 2 New card with fewer ports

A card is replaced by a similar card with fewer ports.

Procedure:

Change the card and put the system into operation again. Similar procedure as described in "Replacing a defective interface card", page 206.

The following data is deleted:

- The system and terminal configuration data of the terminals on the terminalinterfaces that are no longer present in the new configuration.
- The system configuration data of the network interfaces that are no longer present in the new configuration.

Tab. 88 Example: Reducing the number of terminal or network interfaces

16DSI → 8DSI	The configuration data of terminal interfaces 9...16 are deleted.
8BRI → 4BRI	The configuration data of network interfaces 5...8 are deleted.

Note:
If the terminal configuration data of system phones is deleted following the reconfiguration of a card, a warning message will appear beforehand to give you the possibility of cancelling the process. However, this is possible only if the configuration data of the original card was not already deleted beforehand.

6. 3. 3. 3 New card with more ports

A card is replaced by a similar card with more ports.

Procedure:

1. Change the card and put the system into operation again. Similar procedure as described in "Replacing a defective interface card", page 206.
2. In the WebAdmin view *Cards and modules* (=4g) *Confirm* the new cards.
3. Configure new ports.

The system configuration data (User No., User configuration, etc.) of the terminals on the new ports is created as new data (default values).

Tab. 89 Example: Expanding the number of terminal or network interfaces

8DSI → 16DSI	The configuration data of terminal interfaces 9...16 is created as new data.
4BRI → 8BRI	The configuration data of network interfaces 5...8 is created as new data.

6. 3. 3. 4 Change slot

Interface cards can be moved to a different slot. The terminal configuration data of the system phones can be transferred.

Procedure:

1. Change the slot and put the system into operation again. Similar procedure as described in "Replacing a defective interface card", page 206.
2. Connect the system phones to the ports of the new slot.
3. Reconfigure the port assignment.
4. In the WebAdmin view *Cards and modules* (=4g) *Confirm* card in the new slot and *Delete* it from the old slot. The configuration data at the old slot location is now deleted.

Note:
Not all cards can be equipped on all slots (see "Component mounting rules", page 115).

6. 3. 4 System modules

The category system modules comprises the modules expandable as an option (DSP modules, IP media modules, call charge modules) and the mandatory modules (RAM module).

6. 3. 4. 1 Change DSP module

DSP modules are available in various versions (SM-DSPX1, SM-DSPX2, SM-DSP1, SM-DSP2). Compared with DSP modules, modules with the designation DSPX are fitted with more powerful DSP chips. The following describes how to replace a DSP module if it is defective or how to replace it for a more powerful module. DSP modules are fitted to the call manager card.

To change a DSP module, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the old or defective module by loosening the fastening screw and carefully pulling the module out vertically of the module slot.

   Note:
   If multiple modules are equipped and the defective module is not topmost, the spacing sleeves have to be loosened and the modules pulled. The order of the modules on the slot is relevant only if different types of modules are equipped.
4. Press the new module downward evenly on both connectors to the stop.
5. Secure the module with the fastening screw.
6. Carefully push back the call manager card into the shaft and gently press the card as far as it goes into the connection on the backplane.
7. Secure the call manager card back into its slot with the screw.
8. Restart the call manager by pressing the On/Off button on the call manager card.

6. 3. 4. 2 Changing the IP media module

IP Media modules are fitted either to the call manager card or to PRI trunk cards.

To replace a defective IP media module to a call manager card, proceed as follows:

> ⚠ **CAUTION!**
> Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the defective module by loosening the 2 fastening screws and carefully pulling the module out vertically of the module slot.
4. Place the new module in the slot and press it down evenly into the slot as far as the stop.
5. Fit the module on to the call manager card from below using the 2 fastening screws.
6. Carefully push back the call manager card into the shaft and gently press the card as far as it goes into the connection on the backplane.
7. Secure the call manager card back into its slot with the screw.
8. Restart the call manager by pressing the On/Off button on the call manager card.

Proceed accordingly to replace one defective IP media module to a PRI trunk card.

6. 3. 4. 3 Replacing the call charge module

Call charge modules are fitted to FXO trunk cards.

To replace a defective call charge module on an FXO trunk card, proceed as follows:

> ⚠ **CAUTION!**
> Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the FXO card and remove the card by pulling the fastening screw.
3. Remove the defective module by loosening the fastening screw and carefully pulling the module out vertically of the module slot.
4. Place the new module in the slot and press it down evenly into the slot as far as the stop.
5. Secure the module with the fastening screw on the spacer sleeve.
6. Carefully slide the FXO card into the slot shaft and gently press the card as far as it goes into the connection on the backplane.
7. Use the screw to secure the FXO card back into its slot.
8. Restart the call manager by pressing the On/Off button on the call manager card.

6. 3. 4. 4 Changing the RAM module

The RAM module is fitted to the call manager card and available as a spare part.

To replace a defective RAM module, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the defective module by pressing the two lateral metal clamps outward at the same time and gently lifting the module.
4. Place the module at a slight angle into the slot (see Fig. 79).
5. Carefully press the module downwards until the two lateral metal clamps engage.
6. Carefully push back the call manager card into the shaft and gently press the card as far as it goes into the connection on the backplane.
7. Secure the call manager card back into its slot with the screw.
8. Restart the call manager by pressing the On/Off button on the call manager card.

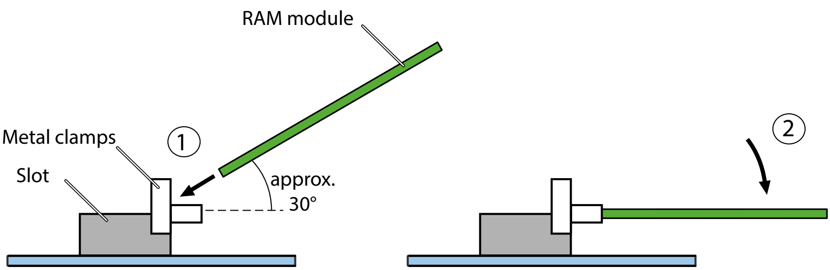


Fig. 79 Changing the RAM module

6. 3. 5 System cards

The category system cards comprises the EIM card and the Flash card.

6. 3. 5. 1 Replacing the EIM card

The EIM card is located in a chip-card holder that is secured directly on the call manager card. The position of the chip-card holder on the call manager card is shown in Fig. 78.

To fit an EIM card, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Lift the EIM card slightly at its bevelled corner, and slide it out of the chip-card holder by gently pushing the guide tongues.
4. Push the new EIM card under the guide tongues and through to the stop in the chip-card holder. Make sure the contacts of the EIM card are facing downwards and the bevelled edge of the EIM card is pointing towards the edge of the call manager card and not against the capacitor (C) (see Fig. 80).
5. Carefully push back the call manager card into the shaft and gently press the card as far as it goes into the connection on the backplane.
6. Secure the call manager card back into its slot with the screw.
7. Restart the call manager by pressing the On/Off button on the call manager card.

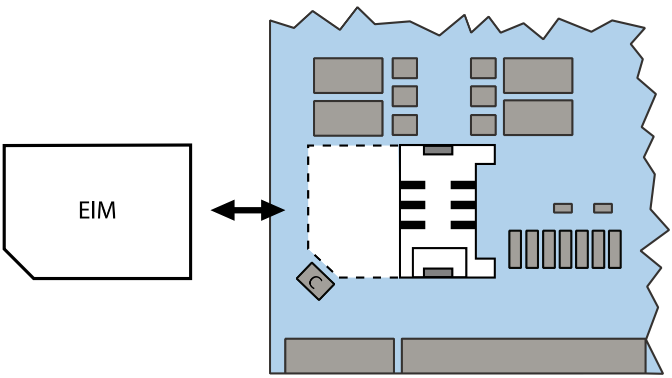


Fig. 80 EIM card

Notes:
- The EIM card must be fitted before the system is put into operation. The communication server will not start without the EIM card.
- If the defective EIM card was replaced by a new one, all DECT cordless phones must be logged on again. This is necessary because the DECT identification numbers are stored on the EIM card.

6. 3. 5. 2 Replacing the Flash Card

The Flash card is fitted to the call manager card and available as a spare part.

To replace a defective Flash card, proceed as follows:

⚠ **CAUTION!**
Be sure to observe the "Safety regulations", page 98.

1. Carry out preparations (see "Preparations", page 204).
2. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
3. Remove the defective Flash card by pulling it out on the side.
4. Fit the new Flash card and gently press the card as far as it will go into the plug-in connection.
5. Carefully push back the call manager card into the shaft and gently press the card as far as it goes into the connection on the backplane.
6. Secure the call manager card back into its slot with the screw.
7. Restart the call manager by pressing the On/Off button on the call manager card.

Notes:
- The Flash cards are expected to meet high demands in terms of data security (read and write cycles). That is why only original Flash cards are to be used.
- Flash cards that are ordered as spare parts do not contain any software. In this case an Emergency Upload has to be carried out (see "Loading new or older system software with System Search", page 201).

6. 3. 6 Call manager card CPU1

If the components on the call manager card are defective or permanently faulty, the entire call manager card must be replaced. As a spare part the call manager card does not contain any RAM module, Flash card or EID card. They can be taken from the defective call manager card and fitted to the new call manager card.

To replace a call manager card, proceed as follows:

⚠ **CAUTION!**
Be sure to observe the "Safety regulations", page 98.

1. Back up the configuration data and audio data, if still possible.
2. Carry out the preliminary steps if still possible (see "Preparations", page 204).
   Note: If the call manager cannot be shut down in the normal way, its shut-down has to be forced (see "Call-Manager display and control panel", page 218).
3. Unscrew the screw on the call manager card and remove the card by pulling the fastening screw.
4. Replace the system modules (see "System modules", page 208), the system cards (see "System cards", page 210) on the new call manager card.

5. Dismantle all the connected cables in such a way that the new communication server can be identically reconnected.
   Note: The CPU card is not dismantled but replaced complete with metal housing.
6. The new communication server can now be reassembled, fitted and installed in the reverse sequence.
7. Restart the call manager by pressing the On/Off button on the call manager card.
8. Carry out a first start of the system (see "First start via WebAdmin", page 192) and upload the configuration data from a backup file back on to the communication server.

Tip:
A defective call manager card may make it impossible to read out unstored configuration data. In such cases the data can be saved using a new call manager card by replacing the Flash card.

6. 3. 7 Applications card CPU2-S

If chips on the applications card are defective or permanently faulty, you need to replace the entire applications card.

To replace an application card, proceed as follows:

CAUTION!
Be sure to observe the "Safety regulations", page 98.

1. Shut down the application server via the control panel (see "On/Off key", page 218).
2. Detach the cables of any assigned interfaces on the front panel of the applications card.
3. Unscrew the screw on the applications card and remove the card by pulling the fastening screw.
4. Carefully slide the new application card into the shaft of slot 2 and gently press the card as far as it goes into the connection on the backplane.
5. Use the screw to secure the card in its slot.
6. Connect the cables of any assigned interfaces on the front panel of the applications card.
7. Start up the applications server by pressing the On/Off button on the applications card.

See also:
For more information about installing, configuring and upgrading the software of the CPU2-Sapplication card, see the CPU2-S application card installation manual.

6. 3. 8 Replacing system terminals

6. 3. 8. 1 DSI system phones

Phones with the same level of added features

Replacing a defective phone

Once the defective DSI system phone has been replaced by an identical phone the terminal configuration data is automatically transferred.

Relocating a phone

The assigned port can be modified in the terminal configuration via WebAdmin, and the phone connected on the new slot. The terminal configuration data is preserved.

Phones with a different level of added features

If a phone is replaced with another type of phone, most of the terminal configuration data can be taken over using *Multi edit*. A separate *Multi edit (keys)* function is available for the key configuration. Details can be found in the WebAdmin help for the view *Standard terminals* (*=qd*).

6. 3. 8. 2 DECT terminals

Replacing a radio unit

1. Dismantle the defective radio unit.
2. Fit the new radio unit.

Note:
If the ports of a radio unit are to be changed or if a radio unit is no longer used, it is important to remove the radio unit in the system configuration. If not, start-up problems may occur when another radio unit is connected to the same ports.

Replacing a cordless phone (a phone without microSD card)

1. Cancel the registration of the old cordless phone.
2. Register the new cordless phone. The cordless phone data is preserved until the user number is also deleted.

Cancelling the registration of a cordless phone on the system

In WebAdmin in the edit view of the cordless phone, click *Cancel registration*.

Tip:
The identification of the cordless phone is deleted only if the cordless phone is located within the coverage range of a radio unit; otherwise, it must be deleted manually on the cordless phone (see the cordless phone's User's Guide). The user number and data in the system are retained.

Registering a cordless phone on the system

1. Prepare the cordless phone for registration (see the cordless phone's User's Guide).
2. Prepare system for registration. In WebAdmin in the edit view of the cordless phone, click *Register*.

Note:
With some phone types, the user of the cordless phone may have to identify himself to the system using an authentication code (AC). This authentication code is issued after the *Register* button is clicked.

Replacing a cordless phone (a phone with microSD card)[1)]

The special microSD card is suitable for replacement with wireless DECT phones Mitel 620/622 DECT, Mitel 630/632 DECT and Mitel 650 DECT. The card stores the cordless phone's registration data on the communication server and the most important local settings. This guarantees that in case of device defect - by taking the card along - the operation on a replacement device can be continued within a short period and without re-registering.

Each card (like each cordless phone) has its own, globally known unique serial number for DECT devices (IPEI: International Portable Equipment Identity), used for the registration process on DECT communication systems. In an operation with the card, the data stored on the card is always used.

Notes:

- The microSD card can only be used as from Device hardware 2 (concerns Mitel 620 DECT, Mitel 630 DECT).
- Use the card only after reading this detailed description of the card functions. Failing to observe these recommendations may cancel the registration of operational devices.
- All registration and device data on the card is encrypted and protected against copying.
- Do not use the card with other devices (e.g. camera) to avoid reformatting the card mistakenly and to have enough storage space.
- The card can no longer be used with the cordless phones after being erased or formatted.

1)Supported as of R2.1

– Commercially available microSD cards cannot be used (except to copy local settings, see page 217).

Using a microSD card

Note:
The microSD card must be handled very carefully. The contacts must be free from dust, humidity, oil, etc. Do not store the card in warm areas (exposed to direct sunlight, for example). Do not bend the card as this may damage the contacts.

1. Switch off the cordless phone.
2. Open the battery compartment and remove the battery.
3. Push the card holder downwards and carefully tilt the cover sligghtly upwards (see Fig. 81 on the left).

CAUTION!
Never touch the now visible and shining golden contacts! Static discharges may lead to device malfunction.

4. Place the card in the holder (with the contact surfaces downwards and the side card interfaces leftwards).
5. Close the card holder then carefully push it upwards until it snaps into place.
6. Only for Mitel 620 DECT, Mitel 630 DECT with black card holder:
   Take the protective cover provided with the card and put it on top of the card holder (see Fig. 81 rightwards).

   Note:
   The protective cover should not be used for Mitel 620 DECT, Mitel 630 DECT with a white card holder or in Mitel 622 DECT, Mitel 632 DECT and Mitel 650 DECT.

7. Insert the battery and cover the battery compartment.

Fig. 81 microSD card

Behaviour after inserting a new microSD card

After starting the cordless phone you will receive, in the start phase, a message informing you that a new card has been detected. The two typical cases are described below:

Cordless phone has not yet been registered:

Accept the new card.
→ The local settings are copied to the card.

Register the phone on the communication server.
→ The registration data is stored on the card.
→ Modifications on the local settings are henceforth also stored on the card.

Cordless phone is already registered:

Accept the new card.
→ The local settings are copied to the card.
→ The registration data is copied to the card and erased from the cordless phone memory.
→ Modifications on the local settings are henceforth also stored on the card.

Behaviour after inserting a valid microSD card

After starting the cordless phone you will receive, in the start phase, a message informing you that a new card with a new ID has been detected.

Accept the card.
→ The cordless phone restarts.
→ The card's registration data and local settings are used.
→ The original data remains stored in the cordless phone and is reactivated once the card is removed.

Copying local settings using a commercially available microSD card

This procedure is helpful if several cordless phones with the same local settings must be preconfigured.

1. Carry out on a master cordless phone without microSD card the local settings you want.
2. Switch off the master cordless phone, insert a commercially available microSD card then restart the master cordless phone.
3. Confirm the information that the microSD card is invalid.
4. Select *Menu* - *Settings* - *General* - *Administration* - *Diagnostics* - *File Mgmt. Device* then copy all user data to the microSD card.
   -> The card is now specially marked as a copy card.
5. Switch off the master cordless phone, remove the card and insert the card in the target cordless phone to which the data must be copied.

6. Start the target cordless phone and confirm the information that the user data on the card will be used.
7. Copy all user data from the card to the memory of the target cordless phone.
   -> The target cordless phone restarts.
8. Switch off the target cordless phone and remove the card.
   -> After the target cordless phone is switched on again the copied user data is used.

6. 4 Call-Manager display and control panel

The display and control panel on the call manager card consists of the colour display with the navigation keys and the On/Off button with integrated status LED. It is used to indicate operating states and carry out functions.

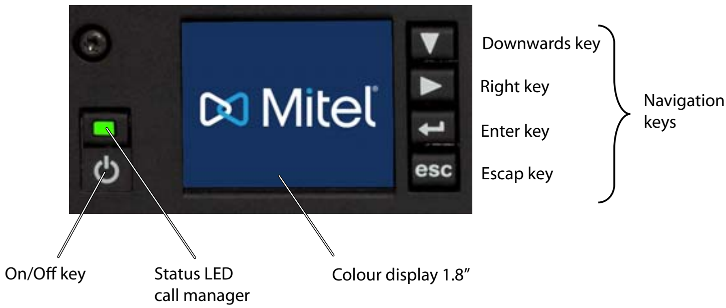


Fig. 82 Mitel 470 display and control panel

6. 4. 1 PIN control panel

A number of functions executed via the navigation keys require a PIN (e.g. run first start).

The PIN always consists of 4 digits and can be modified via *SystemUserInterface* user account:

Tab. 90 Default PIN control panel

Default PIN	4321

It is advisable to change the PIN immediately to prevent unauthorized access to the communication server.

6. 4. 2 On/Off key

Pressing the On/Off button starts up the call manager (which is switched off).

In normal operation a short key press of the On/Off key brings up the Shut Down menu, offering the choice of shutting down the Call Manager, the application server or the entire communication server. The navigation keys are used to select from the menu.

Tab. 91 On/Off key

Function	Action	Note
Start the call manager	Short key press	Requirements: • Power supply on • Executable system software loaded
Shut down the communication server, call manager or applications server	Short key press	The display shows the Shut Down menu with the following selection: • Shut down full system: Shut down communication server (CPU1 and CPU1[1]).[2] • Shut down Call Manager: Shut down CPU1 only • Shut down Application Server: Shut down CPU2[1] only
Force Call Manager shut down	Keypress longer than 6 seconds	Note: The forced shut-down of the Call Manager should only be made if shutting down via the Shut Down menu is no longer possible for whatever reason.

1) Shutting down the applications server can take some time and can be checked using the status LED on the On/Off button (see Tab. 98).

2) This corresponds to the "Off state" in accordance with EU Directive 2005/32/EC.

Notes:

Never disconnect the communication server from the power supply to trigger a restart. This can result in data losses and prevent a restart.

Tips

– The shut-down menu can also be used via the Call Managers control panel. A restart menu is also available, in which CPU1 and CPU2 can be restarted separately.
– CPU1 and CPU2 can also be restarted via WebAdmin.

6. 4. 3 Status LED

Status LEDs can be found on the On/Off buttons and on the Ethernet interfaces of the call manager card.

The status LED on the On/Off button of the call manager is used as an operating state and error indicator during the start-up phase and during operation.

The status LED may be lit in the three colours green (G), orange (O) and red (R), flashing slowly or rapidly, or be inactive (–).

An LED activation period lasts 1 second and is subdivided into 4 units of 250 ms. Different display patterns can be displayed in this way.

Tab. 92 Examples of display patterns

LED activation period ← 1 s →				LED	Description
On	On	On	On	G	LED lit green
On	On	Off	Off	O –	LED slowly flashing orange
On	Off	On	Off	O – R –	LED flashing rapidly orange/red

6. 4. 3. 1 Startup and operating state display

In the system setup the status LED indicates the current operating state of the Call Manager.

The start-up phase can be divided into three parts:

System setup 0:
In this phase, the system can be set to the boot mode (see "Boot mode", page 220)

System setup 1:
The colour display is not yet operational. Any errors that occur are indicated with the status LED (see "Error display with status LED", page 221).

System setup 2:
The colour display is operational. In this phase, the boot menu is shown (see "Boot menu", page 221). Any errors that occur are displayed via the colour display.

Tab. 93 Display pattern at system setup

Pattern	LED	Duration [s]	Meaning	Start-up phase
0	–	steady	Call manager is switched off	
1	R	~1,5	Red LED test	0
2	O	~1,5	Orange LED test	0
3	G	~1,5	Green LED test	0
4	G – G	~4	RAM test, load boot software, boot software CRC test	1
5	O –	~10	Boot software running, load system software, system software CRC test	2
6	G –	steady	System software running error-free	

6. 4. 3. 2 Boot mode

The boot mode enables an Emergency Upload via the Ethernet interface (EUL via LAN). This is required whenever there is no longer any executable system software stored on the communication server for whatever reason.

The boot mode is indicated by the status LED flashing red.

Tab. 94 Display pattern in the boot mode

Pattern	LED	Duration	Meaning
10	R –	As long as the boot mode is active	Boot mode active

To access the boot mode press the enter key during the LED test red, which is executed during the start-up phase 0. After a wait time of approx. 10 seconds, Pattern 10 is displayed. A short while later, "BOOT MODE ENTERED" is displayed.

The boot mode remains active until the Emergency Upload is completed or the system is restarted manually.

6. 4. 3. 3 Error display with status LED

Errors that occur during the start-up phase1 are indicated with the status LED.

Tab. 95 Error displays during system setup 1:

Pattern	LED	Duration	Meaning
7	O-O-	As long as the error remains	RAM test faulty
8	O-R-	As long as the error remains	Boot software missing
9	R-R-	As long as the error remains	CRC test boot software faulty

6. 4. 3. 4 Boot menu

The boot menu is shown during the start-up phase 2 (LED pattern5 in Tab. 93) for approx. 3 seconds. The boot menu allows the user to reset the IP address data or to carry out a first start. The boot mode is exited automatically and the startup then continues normally if no input is made within 3 seconds.

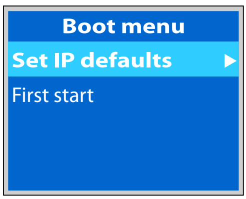


Fig. 83 Boot menu Mitel 470

6. 4. 3. 5 Display of event messages

If an event message occurs in normal operation, the LED pattern switches from "slowly flashing green" to "slowly flashing orange-green" and the event message is indicated on the colour display.

Tab. 96 Display of event messages in normal operation:

Pattern	LED	Duration	Meaning
11	O G	As long as the event message exists	Event message present

6. 4. 3. 6 Status LEDs on Ethernet interfaces

For explanations of the status LEDs on Ethernet interfaces see "Status LED", page 161.

6. 4. 4 Colour display

The colour display has different display modes, which depend in part on the Call Manager's operating mode.

The table below summarises the display modes.

Tab. 97 Operating modes and display priorities

Display mode of the colour display	Call Manager operating mode	Trigger event and purpose
Error mode (Error mode)	System setup 2	• Triggered by software or hardware error. • The error is shown on the display. • The system is unable to run.
Boot menu (Boot command mode)	System setup 2	• Is shown during the start-up phase 2 (LED pattern5 in Tab. 93) for approx. 3 seconds. • Allows the user to reset the IP address data or to carry out a first start.
Menu mode (Application command mode)	Normal operation	• Triggered by pressing any navigation key briefly in the traffic load mode. • Allows the user to run various advanced functions.
Traffic load mode (Traffic mode)	Normal operation	• After the startup of the Call Manager or after exiting the menu, idle or event message mode. • Shows the current traffic load of the Call Manager.
Idle mode (Idle mode)	Normal operation	• After a certain amount of time without user interaction from the traffic mode or the event message mode. • Screen saver and energy saving function.
Event message mode (Event message mode)	Normal operation	• After one or more event messages are received.

6. 5 Application server display and control panel

The application server display and control panel consists of one On / Off button and a few status LEDs.

6. 5. 1 On/Off key

Pressing the On/Off button starts up the application server (which is switched off). In normal operation mode, the application server is shut down by briefly pressing the On/Off button.

Notes:
- The application server can also be shut down and started via the Call manager control panel or via WebAdmin in the *Maintenance* / *System reset* (*=4e*) view.
- Shutting down the applications server can take some time and can be checked using the status LED on the On/Off button (see Tab. 98).
- If regular shut-down is not possible (for instance because the application is no longer reacting), the application card is forced to shut down after 2 minutes without the operating system being shut down normally. Unsaved data will be deleted.

6. 5. 2 Status LEDs

Status LEDs can be found on the On/Off buttons and on the Ethernet interfaces. There is also one LED for the USB ports and the hard disk.

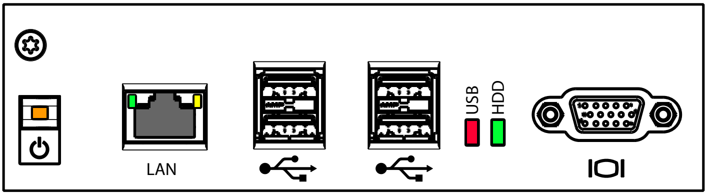


Fig. 84 Status LEDs on the applications server

Tab. 98 Explanation of the status LEDs on the applications card

LED	Signalling	Meaning
On/Off	Steady green	Applications server running fault-free
On/Off	Steady red	Error on the applications server
On/Off	Steady orange	Applications server is switched off
HDD	Flickering green	Hard disk access
USB	Steady red	Power overload on one of the USB interfaces. Note: The maximum permissible current input at the USB interfaces varies (see Tab. 28).
LAN	The Ethernet interface on the applications server is covered as there is currently no provision for its use.	

6. 6 Operations supervision

6. 6. 1 Event message concept

The system generates an event message every time an event or error occurs. The event tables are used to specify how often an event message of a particular type may be generated by the system over a given period before the event message is sent to the allocated signal destinations.

There are 7 event tables that can be allocated to 8 signal destinations:

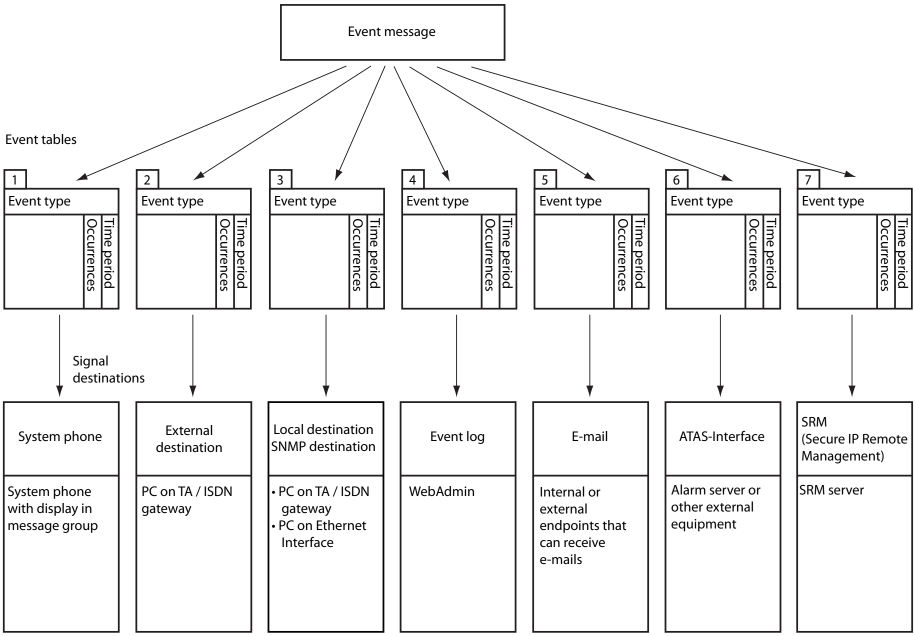


Fig. 85 **Distribution principle for an event message**

6. 6. 1. 1 Event types

Event messages have a certain severity level: *Normal* (blue), *Serious* (yellow) and *Critical* (red). Many event messages have both a negative impact (error occurred) and a positive impact (error corrected). Some event messages have no impact and, thus, no match. Severity level, positive or negative impact (if any) and the information, if there is a match or not, are indicated in the event table.

If an SRM server is indicated as signal destination, the event message severity level results in a change of system status. This can be seen in the SRM agent and is displayed with the corresponding colour (see also section "SRM destination", page 248).

Tab. 99 Event types, in alphabetical order

Event message	Trigger condition	Details[1]	Severity
ATAS: Connection established	ATAS: connection (re) established	Date, time	critical (positive, with match)
ATAS: Connection lost	ATAS: connection lost	Cause (0: Logoff, 1: missing cycle signal), date, time	critical (negative, with match)
BluStar Client back within the licence limit	A sufficient number of licences is now available again for BluStar clients. Parameter 1: 0 (not used) Licence type: 0 and 1: (not used), 2: BluStar CTI, 3: BluStar Softphone, 4: BluStar Video option, 5: BluStar Presence option	Parameter 1, licence type, total purchased licences, date, time	Serious (positive, with match)
Card in service	A card that was previously out of service is back in service again.	Number of the expansion slot, date, time	critical (positive, with match)
Card out of service	A card previously in operation has stopped functioning.	Number of the expansion slot, date, time	critical (negative, with match)
Card reset	A reset was carried out for one card	Number of the expansion slot, date, time	Serious (without match)
Charge counter overflow	Individual cumulative counter or cost centre counter overflow	Cause (0: User / 1: Cost centre / 2: Exchange line / 3: Room), number, date, time	Serious (without match)
CL printer available again	Printout on the system printer available once again	Date, time	Serious (positive, with match)
CL printer blocked	• No response from system printer for past 4 minutes • Printer out of paper or switched off	Interface, interfaces/card number, port number, date, time	Serious (negative, with match)
Compatible PMS application	The external hotel management system (PMS application) is suitable for communicating with the communication server.	Date, time	critical (positive, with match)
Configuration template available	The missing configuration template for a Mitel SIP terminal is now available in the communication server file system.	Date, time	Serious (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
Connection to IP remote management (SRM) failed	IP remote management connection set up (SRM = Secure IP Remote Management) has failed. Cause parameter: 1: Connection attempt failed, 2: Authentication failed, 3: File upload rejected	Cause, date, time	Normal (negative, with match)
Connection to IP remote management (SRM) restored	IP remote management connection has been (SRM = Secure IP Remote Management) successfully restored.	Date, time	Normal (positive, with match)
Connection to PMS system established	A connection with a hotel management system (PMS system) has now been successfully established.	Date, time	critical (positive, with match)
Connection to PMS system failed	An unsuccessful attempt was made to establish a connection with a hotel management system (PMS system). Reason: 1: Call rejected, 2: Destination unobtainable, 3: Destination busy, 4: Connection timeout, 5: Wrong address, 6: Unknown error	Error, date, time	critical (negative, with match)
CPU2 applications card Data communication out of service	Data communications with the CPU2 applications card have been interrupted for an unusually long period of time (> 1 hour) due to an error (after a Windows update or for other reasons).	Date, time	critical (negative, with match)
CPU2 applications card Data communications back in service	Data communications with theCPU2 applications card have been restored.	Date, time	critical (positive, with match)
Creation instance on backup communication server failed	The backup communication server was unable to create or modify a user or terminal instance with the received configuration data. Note: This event message is generated by the backup communication server.	Instance type (0: User, 1: terminal), user number or terminal ID, date, time	critical (negative, with match)
Creation instance on backup communication server successful	The backup communication server was able (following one or more previous failed attempts) to create or modify a user or terminal instance with the received configuration data. Note: This event message is generated by the backup communication server.	Instance type (0: User, 1: terminal), user number or terminal ID, date, time	critical (positive, with match)
CSTA sessions within the licence limit again	*CSTA Sessions* licences are now available again.	Number of licences, date, time	Serious (positive, with match)
CTI first party Connection established	The first-party link was (re-)established	User number, terminal ID, protocol type (0=ATPC3, 1=CSTA) date, time	critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
CTI first party Connection lost	The first-party link was interrupted because the cycle signal is missing.	User number, terminal ID, protocol type (0=ATPC3, 1=CSTA) date, time	critical (negative, with match)
CTI third party: Connection established	The third-party link was (re-)established	IP address, protocol type (0=ATPC3, 1=CSTA), date, time	critical (positive, with match)
CTI third party: Connection lost	The third-party link was interrupted	Cause (0=Logoff, 1= missing cycle signal), IP address, protocol type (0=ATPC3, 1=CSTA) date, time	critical (negative, with match)
Definitive activation licence missing	The initial temporary activation of the communication server for a certain duration (e.g. 90 days) was started. After this period, the communication server switches to restricted operating mode (see "Restricted operating mode", page 83).	Date, time	critical (negative, with match)
Definitive activation licence now present	A licence file with a definitive activation licence was uploaded.	Date, time	critical (positive, with match)
Dual Homing back within the licence limit	There are now enough licences available for registering SIP phones in the Mitel 6800/6900 SIP series on a backup communication server. Note: This event message is generated by the backup communication server.	Date, time	Serious (positive, with match)
E-mail successfully sent	The system has now successfully sent an e-mail. Meaning of the parameter values in Tab. 100	Cause/action=0000, e-mail client, additional information, date, time	critical (positive, with match)
Emergency call ended	The emergency call has been confirmed by a responsible person.	Date, time	critical (positive, with match)
Emergency call started	An emergency number out of the public emergency number list has been dialled. Note: If an emergency number of the internal numbering plan has been dialled, no event message will be generated.	Dialled number (the first 4 digits), user number, terminal ID (if user number ≠ 0) or trunk group ID (if user number = 0), date, time	critical (negative, with match)
ESME reachable	The LAN connection between the SMSC and the ESME is now available	IP address, date, time	critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
ESME unreachable	The LAN connection between the SMSC and the ESME is interrupted	IP address, date, time	critical (negative, with match)
Ethernet activated again	The overload on the Ethernet interface no longer exists. The interface has been reactivated.	Date, time	Normal (positive, with match)
Ethernet deactivated due to high load	The system has detected an overload on the Ethernet interface. The interface is temporarily deactivated.	Date, time	Normal (negative, with match)
External auxiliary power supply failed (Mitel 470 only)	The external auxiliary power supply to the communication server has failed. If the auxiliary power supply unit has been used for redundant operation, there are no short-term limitations. If the auxiliary power supply unit has been used to increase the power supply, the internal power supply unit overflow must be calculated.	Date, time	Serious (negative, with match)
External auxiliary power supply in service (Mitel 470 only)	The external auxiliary power supply to the communication server is working.	Date, time	Serious (positive, with match)
External event message destination not reachable	External signal destination not automatically reachable	Cause (0: Busy /1: Not available /2:(not used), 2: Barred /3: not defined), date, time	Serious (negative, with match)
External event message destination reachable	External signal destination is now reachable	Date, time	Serious (positive, with match)
Fan failure (Mitel 415/430 and Mitel SMBC only)	The fan is jammed or defective or the connection is no longer making contact. • Parameter = 0: No more fans in operation. –> Risk of overheating: Replace defective fan.	Parameter, date, time	critical (negative, with match)
Fan failure (Mitel 470 only)	The fan is jammed or defective or the connection is no longer making contact. • Parameter 1 = 0: No more fans in operation. –> Risk of overheating: System shut down after 2 minutes. –> Replace both fans. • Parameter 1 = 1: Only one fan left in operation. Parameter 2 = Defective fan number –> System still running with only one fan. –> Replace defective fan.	Parameter 1, parameter 2, date, time	critical (negative, with match)

Event message	Trigger condition	Details[1)]	Severity
Fan in operation (Mitel 415/430 and Mitel SMBC only)	The fan is back in service again after a failure. • Parameter = 0: Fan back in service again.	Parameter, date, time	critical (positive, with match)
Fan in operation (Mitel 470 only)	The fan is back in service again after a failure. • Parameter = 0: A fan is back in service again. • Parameter = 1: Second fan back in service again.	Parameter, date, time	critical (positive, with match)
FIAS command buffer full	The command buffer to the PMS interface is full.	Date, time	critical (negative, with match)
FIAS interface usable again	The command buffer to the PMS interface is back below the critical limit.	Date, time	critical (positive, with match)
Inactive radio unit port	Radio unit not responding Reason: 0: Startup running, 1: Not registered, 2: Various nodes, 3: Port not permitted, 4: Local power supply, 5: Not connected, 6: Port reset, 7: Startup error, 8: Unknown error	Card number, port number, radio unit ID/reason, date, time	Serious (negative, with match)
Incompatible PMS application	The external hotel management system (PMS application) is not suitable for communicating with the communication server.	PMS SW version, PMS interface version, PMS interface driver version, date, time	critical (negative, with match)
Incorrect or no wiring adapter (Mitel 415/430 and Mitel SMBC only)	There is no wiring adapter in the wiring adapter slot or the wiring adapter fitted is unsuitable.	Slot number, date, time	Critical (without match)
Insufficient bandwidth	An user in an AIN is trying to set up a connection and the bandwidth currently available with the WAN link is insufficient.	Link ID, WAN link name, available bandwidth in Kbit/s, date, clock	Serious (without match)
Internal event message destination not reachable	Local output blocked or not available	Cause (0: Busy /1: Not available /2:(not used), 2: Barred /3: not defined), date, time	Serious (negative, with match)
Internal event message destination reachable	Local output available once again	Date, time	Serious (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
Internal power supply unit failed (Mitel 470 only)	The internal power supply unit of the communication server has failed. If the auxiliary power supply unit has been used for redundant operation, there are no short-term limitations. If the auxiliary power supply unit has been used to increase the power supply, the external power supply unit overflow must be calculated.	Date, time	Serious (negative, with match)
Internal power supply unit in service (Mitel 470 only)	The internal power supply unit of the communication server is in service.	Date, time	Serious (positive, with match)
IP address added to the DoS black list	A DoS attack has taken place beyond the maximum configured admissible registration attempts or transactions. The IP address concerned has been included in the black list and will remain blocked for a set period.	IP address, Cause (0: Registration / 1: Too many transactions / 2: No session / 3: modified message), date, time	Serious (negative, with match)
IP address changed: Regenerate TLS certificates	The IP address of the communication server has changed. The TLS certificates have to be regenerated. For terminals downcircuit from a NAT without ALG the public NAT gateway address has to be configured.	Date, time	Serious (without match)
IP address removed from the DoS black list	An IP address added previously due to a DoS (Denial of Service) attack was again removed from the black list and is no longer blocked.	IP address, date, time	Serious (positive, with match)
IP phone: Connection lost	An IP system phone is no longer connected to the communication server.	User number, terminal ID, date, time	Serious (negative, with match)
IP phone: Connection re-established	An IP system phone has re-established the connection to the communication server.	User number, terminal ID, date, time	Serious (positive, with match)
IP system phone licence is now available	A sufficient number of licences is now available again for MiVoice 5361 IP / 5370 IP / 5380 IP.	Date, time	Serious (positive, with match)
Language file download failed	The downloading of a language file via FTP server for an MitelSIP terminal has failed.	Parameter 1: FTP server address, Parameter 2: Language file type and name, date, time	Serious (negative, with match)
Language file download successful	The downloading of a language file via FTP server for an Mitel SIP terminal has been successfully completed.	Parameter 1: FTP server address, Parameter 2: Language file type and name, date, time	Serious (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
LCR on alternative network provider	Automatic switch from primary network provider to secondary network provider using LCR function.	Provider ID, date, time	Normal (without match)
Licence available for configured user (Mitel 470 and Virtual Appliance only)	This event message is generated, if all configured users have a user licence (which was not the case before).	Date, time	Serious (positive, with match)
Licence for integrated mobile/external phone available	A sufficient number of licences is now available again for integrated mobile/external phones.	Date, time	Serious (positive, with match)
Licence for PMS interface available	The *Hospitality PMS Interface* licence or a sufficient number of *Hospitality PMS Rooms* licences are now available.	Date, time	Serious (positive, with match)
Licence invalid, restricted operating mode 4 hours after restart	The system software loaded requires a software release licence. Without this licence the system software's functionality is severely restricted 4 hours after the restart.	Date, time	Serious (without match)
Licence missing for configured user (Mitel 470 and Virtual Appliance only)	This event message is generated, if one or more configured users have no user licence. Note: To avoid a flood of messages this event message is generated only once (the first time a user is created without a user licence)	Date, time	Serious (negative, with match)
Licences for offline operations expired	The maximum period of 36 hours for the temporary licence activation has expired.	Date, time	Critical (without match)
Link to gateway satellite lost (Virtual Appliance only)	The communication server has lost the link to the gateway satellite. Without this link, the communication server switches to restricted operating mode after xx hours.	Number of hours until restricted operating mode, date, time	critical (negative, with match)
Link to gateway satellite restored (Virtual Appliance only)	The communication server has been able to restore the link to the gateway satellite.	Date, time	critical (positive, with match)
Link to the licence server (SLS) has failed (Virtual Appliance only)	It has been impossible for a long time to set up a link to the licence server. The system switches to restricted mode after a variable timeout (max. 72 hours).	Date, time	critical (negative, with match)
Link to the licence server (SLS) has restored (Virtual Appliance only)	It has been possible to restore a link to the licence server.	Date, time	critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
Local supply error on radio unit	Local power supply of a SB-4+ / SB-8 / SB-8ANT radio unit failed or unavailable	Card number, port number, date, time	critical (negative, with match)
Local supply on radio unit available	Local power supply of a SB-4+ / SB-8 / SB-8ANT is now again available	Card number, port number, date, time	critical (positive, with match)
Mains voltage failure	Event message once mains power is restored • Mains power has failed more frequently than entered in the trigger table	Date, time	Serious (without match)
Malfunction	A hardware or software error has occurred. The error ID can help Support to pinpoint the possible cause of the error.	Error ID, date, time	Serious (without match)
MiCollab: Terminal limit has been reached	A MiCollab terminal could not be linked to a user because a limit has been reached (reason). reason = 0: Too much terminals per system reason = 1: Too much terminal per user reason = 2: Too much MiCollab clients per user	User number, reason, date, time	Serious (negative, with match)
MiCollab: Within the terminal limits again	A MiCollab terminal could now be linked to a user because it is within a limit again (reason). reason = 0: Terminals per system OK again reason = 1: Terminal per user OK again reason = 2: MiCollab clients per user OK again	User number, reason, date, time	Serious (positive, with match)
Mitel Dialer within the licence limit again	*Mitel Dialer* user licences are now available again.	Date, time	Serious (positive, with match)
Mitel SIP terminals within the licence limit again	*Mitel SIP Terminals* and *Mitel 8000i Video Options* licences are now available.	Parameter 1=1: *Mitel SIP Terminals* licence, Parameter 2=1: *Mitel 8000i Video Options* licence, date, time	Serious (positive, with match)
Monitor event	Monitor event	Monitor Type, Date, Time	Normal (without match)
No configuration template	A configuration template for a Mitel SIP terminal is missing in the communication server file system. Without the configuration template, no configuration file can be generated for this terminal type.	No configuration template, date, time	Serious (negative, with match)
No DECT DSP channels available	DECT channels on DSP-0x overloaded	Date, time	Normal (without match)

Event message	Trigger condition	Details[1)]	Severity
No DTMF receiver available for integrated mobile/external phones	A permanent DTMF receiver (for detection suffix dialling function codes) could not be assigned to an integrated mobile/external phone with enhanced functionality.	BCS Ref., date, time	Serious (without match)
No other system clone detected (Virtual Appliance only)	The clone detection service on the licence server (SLS cloud) could not find any other clone (system with the same EID) for a long time (24 hours).	Date, time	critical (positive, with match)
No response from network	No answer to Call Setup on BRI-T/PRI interface	Port number of the exchange line circuit, date, time	Normal (without match)
No response from user	No answer to incoming DDI call from user on S bus or DSI	DDI No., date, time	Normal (without match)
Node: Connection lost	A node is not connected to the Master for a certain amount of time (configurable).	Node number, date, time	critical (negative, with match)
Node: Connection re-established	A node is reconnected with the Master for a certain amount of time (configurable) after an interruption.	Node number, date, time	critical (positive, with match)
Not enough licences for integrated mobile/external phones	The connection setup with an integrated mobile/external phone has failed because the number of configured mobile/external phones is greater than the number of licences available. All the integrated mobile/external phones remain blocked until a sufficient number of licences are available.	Number of licences, number of configured mobile/external phones, date, time	Serious (negative, with match)
NTP: Time synchronisation failed	Time synchronization via the NTP server (NTP = Network Time Protocol) has failed.	Date, time	Serious (negative, with match)
NTP: Time synchronisation re-established	Time synchronization via the NTP server (NTP = Network Time Protocol) has been restored.	Date, time	Serious (positive, with match)
Outgoing call rejected	Call rejected by the network • On any line: error code 34 • On required line group: error code 44	Port number of the exchange line circuit, cause, date, time	Normal (without match)
Overheat (Mitel 415/430 and Mitel SMBC only)	The temperature inside the communication server is too high. The appropriate measures must be taken immediately to improve the heat dissipation, e.g. by providing the required clearances, lowering the ambient temperature or installing the fan from the rack-mounting set (Mitel 430 only).	Card number, temperature, date, time	critical (negative, with match)

Event message	Trigger condition	Details[1)]	Severity
Overheat (Mitel 470 only)	The temperature inside the communication server is too high. Appropriate measures must be taken immediately to improve heat dissipation. Measures are automatically adopted, depending on where the overheating occurs: FXO and FXS interface card: • the ports are deactivated in groups of 4 ports. • Once they have cooled down below a defined card-specific value, the ports are automatically reactivated group by group. CPU2 applications card • The card will be completely deactivated. Once it has cooled down below a defined value, the card is automatically reactivated. Internal power supply unit PSU2U or call manager card CPU1: • the communication server will be shut down completely. **Notes:** • To prevent the system from overheating, no more than 30% of the FXS ports should be active simultaneously per 32FXS card and no more than 50 FXS ports per system. • PRI, BRI and DSI cards do not have temperature sensors and are therefore never deactivated due to overheating.	Card number, temperature, date, time	critical (negative, with match)
Overload detected on USB port (CPU2) (Mitel 470 only)	A (current) overload was detected on one of the USB interfaces on the applications card (CPU2). Note: The maximum current input at the USB interfaces varies. (see also Tab. 28)	Date, time	Normal (without match)
Port out of service	A port previously in operation has stopped functioning.	Number of the slot, relevant port number, date, time	Serious (without match)
Possible clone detected for your system (Virtual Appliance only)	The clone detection service on the licence server (SLS cloud) has detected a possible clone (system with the same EID).	Date, time	critical (negative, with match)
QSIG licence limit reached	Maximum number of licensed outgoing connections with QSIG protocol exceeded	Route number, user number, date, time	Serious (without match)
Radio unit port active	The radio unit is responding again	Card number, port number, date, time	Serious (positive, with match)
Register error	• Card not fitted • Card not logged on • Card defective	Card number, date, time	Normal (without match)

Event message	Trigger condition	Details[1)]	Severity
Remote maintenance disabled	Remote maintenance has been deactivated	Date, time	Normal (positive, with match)
Remote maintenance enabled	The remote maintenance has been activated (The report is output unfiltered on local destinations).	Date, time	Normal (negative, with match)
Restart of applications card CPU2 executed	The restart of applications card CPU2 was executed successfully.	Date, time	critical (positive, with match)
Restart of applications card CPU2 required	The system has detected that a manual restart of the applications card CPU2 is required (e. g. for a security update).	Date, time	critical (negative, with match)
Restricted operating mode disabled	Restricted mode could be disabled again.	Date, time	critical (positive, with match)
Restricted operating mode enabled (not valid for Virtual Appliance)	The communication server has switched to restricted mode. Cause: 0: No valid licence	Cause, date, time	critical (negative, with match)
Restricted operating mode enabled (Virtual Appliance only)	The communication server has switched to restricted mode. Cause: 0: No valid licence. 1: Link to gateway satellite lost. 2: Max. duration without link to licence server reached. 3: Your system clone confirmed. 4: Licence check mode mismatch in SLS and MiVo400. 5: Support mode enabled.	Cause, date, time	critical (negative, with match)
Satellites missing after supervision time	After an AIN update (Master and all satellites) some satellites no longer have a connection to the Master.	Total satellites missing, Satellites rolled back, Date, Time	Serious (without match)
Send e-mail failed	The system was unable to send an e-mail because an error occurred. Meaning of the parameter values in Tab. 100	Cause/action, e-mail client, additional information, date, time	critical (negative, with match)
SIMPLE/MSRP back within the licence limit	There are now enough licences available for using the MSRP and/or SIMPLE protocol for users.	Date, time	Serious (positive, with match)
SIP account available	The SIP account has successfully registered with the SIP provider.	Provider, account, date, time	critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
SIP account not available	The SIP account cannot register with the SIP provider for a certain reason (0: Provider unobtainable / 1: no permission). The event is triggered only if the parameter *Registration required* is configured to *Yes*.	Provider, account, date, time	critical (negative, with match)
SMS gateway reachable	External SMS gateway again reachable	Date, time	critical (positive, with match)
SMS gateway unreachable	External SMS gateway unobtainable by network provider or incorrectly configured	Date, time	critical (negative, with match)
Software upgrade IP system phone failed	The software update of an MiVoice 5361 IP / 5370 IP / 5380 IP has failed for the stated reason.	User number, terminal ID, reason, date, time	critical (negative, with match)
Software upgrade IP system phone successful	The software update of an MiVoice 5361 IP / 5370 IP / 5380 IP has now been successfully completed after unsuccessful attempt(s).	User number, terminal ID, date, time	critical (positive, with match)
Software upload	During an upload in system status: • *Update running* • *Supervision running* • *Normal operation*	Parameter 1: • 0: "New communication server software loaded, starting...", • 1: New communication server software crashed, rollback performed • 3: New communication server software started and running well Date, time	Normal (without match)
Standard SIP terminals within the licence limit again	*SIP Terminals* and *Video Terminals* licences are now available.	Parameter 1=1: *SIP Terminals* licence, Parameter 2=1: *Video Terminals* licence, date, time	Serious (positive, with match)
SX-200 call data record management system: Connection established	The connection to the SX-200 call data record management system has been successfully established.	Date, time	critical (positive, with match)
SX-200 call data record management system: Connection lost	The connection to the SX-200 call data record management system has been lost.	Date, time	critical (negative, with match)

Event message	Trigger condition	Details[1)]	Severity
SX-200 hotel management system: Connection established	The connection to the SX-200 hotel management system has been successfully established.	Date, time	critical (positive, with match)
SX-200 hotel management system: Connection lost	The connection to the SX-200 hotel management system has been lost.	Date, time	critical (negative, with match)
SX-200 voice mail management system: Connection established	The connection to the SX-200 voice mail management system has been successfully established.	Date, time	critical (positive, with match)
SX-200 voice mail management system: Connection lost	The connection to the SX-200 voice mail management system has been lost.	Date, time	critical (negative, with match)
Synchronisation loss on trunk	A BRI/PRI interface entered in the clock pool has lost the system clock	Port number, date, time	Serious (negative, with match)
Synchronisation re-established	Synchronization with the network has been restored on at least one BRI/PRI interface.	Date, time	Serious (positive, with match)
Synchronisation with backup communication server failed	The primary communication server was unable to transmit the configuration data to the backup communication server. Note: This event message is generated by the primary communication server.	Backup communication server ID, date, time	critical (negative, with match)
Synchronisation with backup communication server successful	The primary communication server was able (following one or more previous failed attempts) to transmit the configuration data to the backup communication server. Note: This event message is generated by the primary communication server.	Backup communication server ID, date, time	critical (positive, with match)
Synchronization on trunk re-established	A BRI/PRI interface entered in the clock pool has been successfully re-synchronized with the system clock.	Port number, date, time	Serious (positive, with match)

Event message	Trigger condition	Details[1]	Severity
System memory usage below the critical range again	The memory usage in the file system for a specific purpose has again fallen below a defined (*Serious* severity level) or critical (*Critical* severity level) value. Purpose (file type ID): 0: File system, 1: Application, 2: Crash-Log, 3: Monitor-Log, 4: Announcement service, 5: Voice mail, 6: Music on hold, 7: Data backup, 8: Hospitality/Accommodation, 9: User folder	File type ID, memory usage in %, date, time	Serious / Critical (positive, with match)
System memory usage over the critical range	The memory usage in the file system for a specific purpose has exceeded a defined (*Serious* severity level) or critical (*Critical* severity level) value. Purpose (file type ID): 0: File system, 1: Application, 2: Crash-Log, 3: Monitor-Log, 4: Announcement service, 5: Voice mail, 6: Music on hold, 7: Data backup, 8: Hospitality/Accommodation, 9: User folder	File type ID, memory usage in %, date, time	Serious / Critical (negative, with match)
System overload	Network access attempted when all lines are seized or the system is overloaded.	Route number, user number, date, time	Normal (without match)
System phone in service again	A system phone on the DSI bus is ready for operation again.	Card number, port number, user number, date, time	critical (positive, with match)
System phone out of service	A system phone on the DSI bus is defective or was disconnected.	Card number, port number, user number, date, time	critical (negative, with match)
Temperature within normal range again	Following overheating, the temperature inside the communication server is back in the normal operating range.	Card number, temperature, date, time	critical (positive, with match)
Temporary activation expires on	Reminder of the missing, definitive activation licence following connection set-up with the communication server.	Expiration date [DD.MM.YYYY], date, time	Serious (without match)
Terminal power supply: Overload (Mitel 470 only)	Rated output slightly exceeded for > 4 s. (see also "Overload shut-down", page 94)	Date, time	critical (negative, with match)
Terminal power supply: Shut-down (Mitel 470 only)	Rated output clearly exceeded for 4 s (see also "Overload shut-down", page 94)	Date, time	critical (negative, with match)
Terminal power supply: Switching back on (Mitel 470 only)	The power supply to the terminals was switched back on after deactivation due to overflow.	Date, time	critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
Terminal power supply: Within normal range again (Mitel 470 only)	The power supply to the terminals is back in the normal rated output range following a slight, preceding overflow.	Date, time	critical (positive, with match)
Test event message	The configuration of message destinations can be tested with this event message.	Date, time	Serious (without match)
The communication server has been restarted	The communication server was restarted manually or automatically due to an error.	Date, time	Critical (without match)
The licence limit for BluStar clients has been reached.	A BluStar client was unable to register because there are too few licences for this client type. Parameter 1: 0 (not used) Licence type: 0 and 1: (not used), 2: BluStar CTI, 3: BluStar Softphone, 4: BluStar video option, 5: BluStar Presence option	Parameter 1, licence type, total purchased licences, date, time	Serious (negative, with match)
The licence limit for CSTA sessions has been reached	An application is unable to set up a CSTA session to monitor/check a terminal because there are too few *CSTA Sessions* licences available.	Max. number of licences, date, time	Serious (negative, with match)
The licence limit for Dual Homing has been reached	A SIP phone in the Mitel 6800/6900 SIP series has attempted to register on a backup communication server and not enough licences are available. Note: This event message is generated by the backup communication server.	Date, time	Serious (negative, with match)
The licence limit for Mitel Dialer has been reached	Mitel Dialer could not be linked to a user because too few licences are available.	Total purchased licences, date, time	Serious (negative, with match)
The licence limit for Mitel SIP terminals has been reached	A Mitel SIP terminal is unable to register or use the video functionality because there are too few *Mitel SIP Terminals* or *Mitel 8000i Video Options* licences available.	Parameter 1=1: Missing *Mitel SIP Terminals* licence, Parameter 2=1: Missing *Mitel 8000i Video Options* licence, Parameter 3=3: Max. number of licences, date, time	Serious (negative, with match)
The licence limit for SIMPLE/MSRP has been reached	A third-party application wishes to use the MSRP and/or SIMPLE protocol for a user, but not enough licences are available.	Date, time	Serious (negative, with match)

Event message	Trigger condition	Details[1)]	Severity
The licence limit for standard SIP terminals has been reached	A standard SIP terminal is unable to register or use the video functionality because there are too few *SIP Terminals* or *Video Terminals* licences available.	Parameter 1=1: Missing *SIP Terminals* licence, Parameter 2=1: Missing *Video Terminals* licence, Parameter 3=3: Max. number of licences, date, time	Serious (negative, with match)
TLS certificate expires soon	A TLS certificate for a SIP node or SIP endpoint is about to expire (*Serious* severity level) or has just expired (*Critical* severity level) and needs to be renewed. If the endpoint type is = 0 (Mitel), then is parameter 2 = node ID. If the endpoint type is = 1 (3rd party), then the remaining parameter data contains the first eleven characters of the certificate name.	Type of endpoint (0: Mitel, 1: 3rd party), node ID or certificate name, date, time	Serious / Critical (without match)
TLS certificate update failed	The update of the TLS certificate for an SIP node or SIP endpoint via FTP has failed and needs to be renewed manually. If the endpoint type is = 0 (Mitel), then is parameter 2 = node ID. If the endpoint type is = 1 (3rd party), then the remaining parameter data contains the first eleven characters of the certificate name.	Type of endpoint (0: Mitel, 1: 3rd party), node ID or certificate name, date, time	critical (negative, with match)
TLS certificate update successful	A TLS certificate for a SIP node or SIP endpoint was successfully renewed. If the endpoint type is = 0 (Mitel), then is parameter 2 = node ID. If the endpoint type is = 1 (3rd party), then the remaining parameter data contains the first eleven characters of the certificate name.	Type of endpoint (0: Mitel, 1: 3rd party), node ID or certificate name, date, time	critical (positive, with match)
TLS certificate was generated: Upgrade non-Mitel endpoints now	A TLS certificate has been generated. If generation is manual, the certificate must be imported manually into the Mitel SIP nodes. The certificate must always be imported manually on all non-Mitel nodes and non-Mitel endpoints.	Date, time	Normal (without match)
TLS server certificate: Validation failed	While a TLS connection is established the validation of the certificate of the TLS server failed.	Service, TCP port, reason, date, time	critical (negative, with match)
TLS server certificate: Validation successful	The validation of the certificate of the TLS server was successful.	Service, TCP port, date, time	critical (positive, with match)
Too few FoIP channels	Setting up a fax connection via T.38 failed because no FoIP channel is available.	Available FoIP channels on node	Serious (without match)

Event message	Trigger condition	Details[1)]	Severity
Too few licences for IP system phones	A MiVoice 5361 IP / 5370 IP / 5380 IP was unable to register because there are too few IP system phone licences.	Date, time	Serious (negative, with match)
Too few licences for PMS interface	Either the *Hospitality PMS Interface* licence is missing or the number of *Hospitality PMS Rooms* licences available is insufficient.	Number of licensed rooms, number of configured rooms, date, time	Serious (negative, with match)
Too few VoIP channel licences	Connection setup failed because the licence limit for simultaneously active VoIP channels has been reached.	No. of licensed VoIP channels, Date, Time	Serious (without match)
Too few VoIP channels	An user is trying to set up a connection that requires one or more VoIP channels which are currently not available.	Available VoIP channels on this node, date, time	Normal (without match)
Too many errors with the same ID	An unusual amount of errors (more than 50 per hour) with the same error ID have occurred.	Error ID, date, time	Normal (without match)
Too many event messages	Number of message types exceeds limit entered in the table on: • "Synch. "Synch.loss on BRI/PRI" • "Outgoing Call Rejected" • "No response from network"	Date, time	Normal (without match)
Too much user data	System capacity exceeded	Date, time	Critical (without match)
Total synchronization loss	Network synchronisation has failed on all BRI/PRI interfaces	Date, time	Serious (negative, with match)
Trial licence expired	The duration for which a trial licence can be used for a specific feature has expired and there is no valid licence.	Licence ID, date, time	Serious (without match)
USER EVENT MESSAGE	With *77[nnnn] from a terminal	nnnn [0000...99999], user number, date, time	Serious (without match)
User memory usage below the critical range again	The memory usage in the file system for a specific user has again fallen below a defined (*Serious* severity level) or critical (*Critical* severity level) value.	User number, memory usage in %, date, time	Serious / Critical (positive, with match)

Event message	Trigger condition	Details[1)]	Severity
User memory usage over the critical range	The memory usage in the file system for a specific user has exceeded a defined (*Serious* severity level) or critical (*Critical* severity level) value.	User number, memory usage in %, date, time	Serious / Critical (negative, with match)
Wake-up call failed	The room wake-up call was not answered	Room No., date, time	Normal (negative, with match)
Wake-up order confirmed	The room wake-up call has now been answered	Room No., date, time	Normal (positive, with match)

1) The node is also always indicated in an AIN.

Tab. 100 Meaning of the parameter values for the event message ***Send e-mail failed***

	Parameter 1 (XXYY)		Parameter 2:	Parameter 3:
Value	**Reason (XX)**	**Action (YY)[1)]**	**E-mail client**	**Additional info depending on the e-mail client (XXYY)**
00	Not defined	Not defined	Not defined	
01	E-mail memory full	Connection set up to SMTP server	Voice mail	XX: Mailbox ID YY: Message ID
02	SMTP server access data invalid	Extended registration on SMTP server	Auto backup	
03	SMTP client cannot set up a connection to the server	Registration on SMTP server	Call recording	User number
04	Authentication failed	Transmission of e-mail address	Event message	
05	Continuous negative answer from SMTP server	Transmission of e-mail recipient address	Call logging for hospitality	
06	Temporary negative answer from SMTP server	Prepare data transmission	Configuration files	XX: User ID YY: Terminal ID
07	No answer from SMTP server	Data transmission in progress		
08	E-mail attachment not found	End data transmission		
09	Invalid host, domain or IP address on the communication server	Prepare authentication (LOGIN)		
10	E-mail text too long (body)	User name authentication (LOGIN)		
11	E-mail attachment too large	Password authentication (LOGIN)		
12	Format of e-mail attachment not supported	Authentication (PLAIN)		

	Parameter 1 (XXYY)		Parameter 2:	Parameter 3:
Value	Reason (XX)	Action (YY)[1)]	E-mail client	Additional info depending on the e-mail client (XXYY)
13	No e-mail recipient address	Prepare encrypted authentication (CRAM-MD5)		
14	Invalid e-mail recipient address	Encrypted authentication (CRAM-MD5)		
15	Invalid e-mail sender address	Preparing to send next e-mail		

1) Action carried out by the SMTP client at the point when the error occurred.

6. 6. 1. 2 Event tables

Event tables (*=f4*) list all the event messages the system is capable of generating (see Tab.).

There are 7 event tables. After a first start, all event tables are assigned at least one destination. This assignment can be modified in the *Message destinations* ((*=h1*)) view. Each event table can be configured individually. This means it is possible with a filter to decide which event message – if any – should be sent to a particular signal destination either immediately, with a delay or not at all.

- *No event*:
  This type of incoming event messages are **never** sent to the linked destination.
- *Every event*:
  This type of incoming event messages are **all** sent to the linked destination.
- *Custom*:
  With this setting, you can determine how often the event message may appear for each period, until they are sent to the linked destination.
  The *Frequency* of an event message may range between 2 and 20. The *Period* is indicated in hours, ranging between 1 and 672. The longest time period corresponds to 28 days or 4 weeks.

Tab. 101 Example of event table

Event type	Frequency	Time period
Total synchronization loss	10	1

In this example an event message is sent to the message destinations if there is a *Total synchronization loss* event message when the system generates the event message 10 times within 1 hour.

6. 6. 1. 3 Signal destinations

After a first start, all event tables are exactly assigned to a message destination. (Exception: *Local destination* and *SNMP destination* use this event table.) You can assign event tables to several or no message destinations

The destinations are configured in the *Message destinations* (*=h1*) view.

Signal destination system phone 1 and 2

Event messages are sent to all system phones with display and entered in the corresponding message group.

- Destination system phone 1:
  - By default allocated to event table 1, which is preconfigured for common use.
  - Fix allocated to message group 16.
- Destination system phone 2:
  - By default allocated to event table 8, which is preconfigured for front desk terminals in hospitality environments.
  - Fix allocated to message group 15.

External signal destinations

Depending on the event table allocated, event messages (normally Table 2) are sent to a specified external signal destination. Two external signal destinations can be specified:

- 1 primary external signal destination
- 1 alternative external signal destination

If the system issues an event message, the event message opens a PPP communication channel from the public network of the communication server to a terminal adapter or modem. Once the event message has been confirmed, the system clears down the PPP connection.

Signalling an event message to an external signal destination

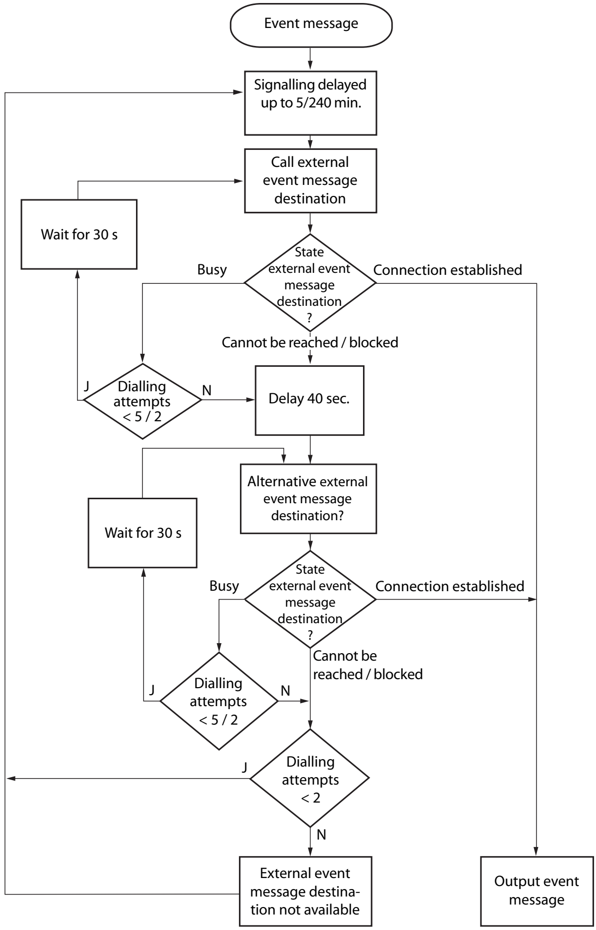


Fig. 86 Flowchart of the signalling of an event message to an external signal destination

The following principles govern the way event messages are signalled to an external signal destination:

- Individual event messages are not signalled if they occur at short intervals. The event messages are stored temporarily for 5 minutes and then sent together to the external signal destination.
- If over a period of one hour an attempt is made unsuccessfully to send the event messages to the external signal destination, the signalling period is extended from 5

minutes to 4 hours. As soon as the event messages are successfully output at the external signal destination, the time period is reset to 5 minutes.

- If over a period of 1 hour an attempt is made unsuccessfully to send an event message to an external signal destination, the number of dialling attempts is reduced from 5 to 2. As soon as an event message has been successfully sent, the number of dialling attempts is increased to 5 again.
- If the attempt to send an event message to an external signal destination was unsuccessful, the system will generate the event message *External event message destination missing*.

Note:
Event tables and signal destinations should be set in such a way that the event message *External event message destination missing* is signalled immediately to any signal destination still available.

Local signal destinations

Depending on the event table allocated, event messages (normally Table 3) are sent to a specified local signal destination.

PPP links:
Like with an external signal destination the event message opens a PPP communication channel from the communication server to a terminal adapter or modem. Once the event has been confirmed, the system clears down the PPP connection.

Ethernet link:
A PC connected either directly to the Ethernet interface or to the communication server via a LAN can be configured as the local signal destination.

Notes:

– The local destination is linked with the same event table as the SNMP destination. Any changes to the link and/or filter criteria for the linked event table also apply to the SNMP destination.
– Event tables and signal destinations should be set in such a way that the event message *External event message destination missing* is signalled immediately to any signal destination still available.

SNMP destination

Depending on the event table allocated, event messages (normally Table 3) are sent to a specified SNMP destinations.

SNMP stands for "Simple Network Management Protocol" and is used by Network Management Systems (NMS).

If the Network Management System is to know the potential events of the communication system, the corresponding system components have to be defined in the form of configurable objects (Managed Objects: MO). These objects and the related event messages are stored in an object library referred to as the Management Information Base (MIB).

You will find the interface description and the different MIB versions on Mitel InfoChannel – Mitel Solution Alliance - API and Interface Information - MiVoice Office 400 - MiVoice Office 400 Network Management.

To access these documents, you have to be a member in Mitel Solution Alliance (MSA). If you are not a member yet, go to Mitel website and search for “Mitel Solution Alliance” where you can join. A membership on level MSA partner (MP) is sufficient.

5 SNMP destinations can be defined. Forwarding to the SNMP destinations can be activated and deactivated independently of the forwarding to the local and external signal destinations.

Notes:
The SNMP destination is linked with the same event table as the local destination. Any changes to the link and/or filter criteria for the linked event table also apply to the local destination.

Signal destination event log

Normally, the signal destination event log is assigned to Event table 4. The filter on this event table is preconfigured for most event types in such a way that event messages are entered in the event log once they arrive.

If the signal destination event log is assigned a different event table or if event table 4 is reconfigured, the event messages are entered in the event log in accordance with the new event table or the new configuration.

The last 254 event messages are recorded in the *Event log* (*=r5*). *Active event messages* (*=mr*) and the last 10 *Power failures* (*=bn*) are recorded in separate logs.

If the maximum number of entries is exceeded, the oldest entry in each case is deleted.

If active event messages are available, they are indicated in WebAdmin on the left, with the symbol.

E-mail signal destination

With the e-mail client integrated in the communication server, event messages can be sent to internal or external e-mail destinations. Normally, the signal destination *E-mail destination* is automatically assigned to event table 5. Up to 5 e-mail destinations can be defined, and e-mail notification can be activated or deactivated globally.

For the communication server to send the e-mails the access to the e-mail service provider's SMTP server must be configured in the *SMTP server* (*=rm*) view.

Destination alarm server (ATAS)

Event messages can also be sent via the ATAS interface, for instance, to an alarm server. This may be an Mitel Alarm Server or a third-party alarm server. The use of the ATAS protocol is subject to a licence.

After a first-start of the communication server, the signal destination *Alarm server (ATAS)* is automatically allocated event table 6. The notification service via the ATAS interface to the alarm server can be globally switched on or off.

SRM destination

Event messages can also be sent to the SRM server. Depending on the severity level in the SRM agent, this changes the system status on the corresponding communication server line. The line colour changes at the same time. If the corresponding positive event message arrives later or if the event message is confirmed in WebAdmin, the status and colour are restored again. The following system statuses are defined:

- *Normal* (Blue colour):
  No active event messages with the severity level *Serious* or *Critical* is available.
- *Serious* (Yellow colour):
  At least one event message is available and needs to be closely examined. (Example: *Charge counter overflow*)
- *Critical* (Red colour)
  At least one event message is available and is hampering the system's function. (Example: *Fan failure*)

Note:
Not all negative event messages have a positive match. In this case, the event messages must be confirmed manually in WebAdmin.
Event messages, which are not *Serious* or *Critical*, are not sent to the SRM server. The severity of individual event messages is given in the Tab. 99 table.

Example:
Power output: There are no serious or critical event messages. The communication server line in the SRM agent is blue and the system status is on *Normal*.

1. The event message *Charge counter overflow* reaches the SRM server.
   → The communication server's system status in the SRM agent changes to *Serious*, and the destinations turn yellow.
2. The event message *Fan failure* reaches the SRM server.
   → The communication server's system status in the SRM agent changes to *Critical*, and the destinations turn red.
3. The event message *Charge counter overflow* is confirmed in WebAdmin in the *Active event messages view* (*=mr*).

→ The system status of the communication server in the SRM agent remains on *Critical*, and the destinations on red, because there is still an event message with this severity.

4. The event message *Fan failure* reaches the SRM server.
   → The communication server's system status in the SRM agent changes to *Normal*, and the destinations turn red.

After a first-start of the communication server, the *SRM destination* is automatically allocated event table 7. The notification service to the SRM destination can be switched on or off.

On the SRM server the status modification per communication server must be allowed and configurations are also required in WebAdmin. You can find a configuration guide in WebAdmin help under the *Message destinations* *=h1* view.

Testing the signal destination configuration

To test the configuration, a test event message can be separately initiated for each destination in the WebAdmin configuration (*Message destinations* *=h1* view). The event message is signalled without any delay, directly at the selected signal destination.

If the communication server is connected via a modem or terminal adapter, the test event messages will be signalled only once the connection is cleared down.

6. 6. 2 Operating state and error displays

6. 6. 2. 1 System operating state

During the start-up phase, various self-tests are performed and the individual phases are indicated with the status LED on the front panel (see "Status LED", page 219).

When operation is OK, the status LED flashes green, regularly, and once per second in the display on the front panel. The system is in normal operation mode. All additional information and operating modes are indicated using the colour display on the front panel (see "Colour display", page 222).

6. 6. 2. 2 System error displays

Whenever the system detects an error, it displays the corresponding error code in the colour display on the front panel (providing the communication server is still powered and the display is working). During system startup, if the colour display is not yet fully functional, any errors that occur are indicated with the status LED (see "Error display with status LED", page 221).

In the event of sporadic errors check the installation for earth loops.

6. 6. 2. 3 Terminals

Tab. 102 Malfunctions on the terminal side

Error description	Error cause / error handling
Digital system phones on the DSI bus display *Not Configured* along with the node number, the slot number and the port number.	No terminal has yet been created on the connected port or an incorrect terminal selection digit (TSD) has been allocated to the terminal: • Check system and terminal configuration • Check installation and connecting cable
System phones do not obtain any dial tone when seizing a line; display reads *Not available*.	Replace phone or interface card.
Terminals with configurable dialling method experience sporadic malfunctions whenever control key is pressed.	System earth must not be connected on terminals configured for MFV/DTMF (double signalling on Flash/earth key).
Analogue terminals do not obtain a dial tone when off-hook.	No terminal has been created on the connected port or the terminal created has not been allocated to a user. • Create a terminal and allocate a user • Check installation or connecting cable

6. 6. 2. 4 Operating state of the Mitel DECT radio units

Each radio unit is equipped with 3 LEDs. The operating state the radio units is indicated by different colours and flashing sequences in cycles of 1 s, specifically by one of the two outer LEDs on the SB-4+ and by both outer LEDs on the SB-8 / SB-8ANT (separately for each DSI bus). Each character (G, R or -) corresponds to 1/8 of a second.

Example:
During the synchronization phase GGGGRRRR the LED flashes periodically.
1/2 second green, 1/2 second red.

Tab. 103 Flashing sequences of the status LED on the DECT radio unit

State	Cycle	Meaning
No flashing	– – – – – – – –	LED switched off / software not running / RU not connected
Red		Error:
	R R R R R R R –	DSI bus not in order
	R – – – – – – –	Power supply error or DSI line too long

State	Cycle	Meaning
Green / red		Startup process:
	G R R R R R R R	DSI ok
	G R G R G R G R	Software is uploaded
	G G G G G R R R	Synchronizing
	G G G G G G G R	DECT is being started
	G G G G G R G R	HF Power Down / DECT System Status Passive[1)]
Green		Normal operation (requirement: LED not switched off):
	G – – – – – – –	All B channels available
	G G G G – – – –	1 to 3 B channels busy
	G G G G G G G –	3 B channels busy

1) This operating state appears in the following situations:
- During a configuration data upload
- After a system first-start
- If in WebAdmin in the *DECT* view (*=sa*) the parameter *DECT system status* is set to *Passive*.
- If no location area is assigned to a radio unit - (This may happen after adding a radio unit to a system with several Location Areas, which is the case when a radio unit has already been set in a Location Area unequal 0). In this case the added radio unit has to be manually allocated to the selected Location Area.)

An orange status LED indicates that DECT signalling is active, i.e. DECT sequences are currently being transmitted between the cordless phone and the radio unit. Examples:

- With each keystroke on the cordless phone the LED briefly lights up orange.
- During a cordless phone firmware download the orange LED remains lit until the download is completed.

On an SB-8ANT radio unit the middle LED indicates whether the internal or external antennas are active. If the LED is lit green, the external antennas are active.

Note:
After the system initialization the radio unit starts in status "DSI ok". It is only ready to operate once at least one DECT user has been entered in the numbering plan or once in WebAdmin the parameter *DECT system status* has been set to *Active*.

6. 6. 2. 5 Malfunction of the Mitel DECT radio unit

Tab. 104 Malfunction of the Mitel DECT radio unit

Error description	Error cause / error handling
No radio connection in a coverage area.	Check LED on radio unit: LED is flashing red (short red phase): • Check power supply / line length of DSI bus cable LED is flashing red (long red phase): • Check DSI bus cable • Unplug DSI bus cable for one minute, then reconnect LED is flashing green (long green phase): • All B channels busy
Radio unit not activated.	LED on radio unit is flashing red/green (various patterns): • Radio unit in startup phase LED on radio unit is flashing red (long red phase): • Radio unit defective If LED on radio unit not flashing: • Check trunk connections • Radio unit defective • LED of the radio units deactivated throughout the system

6. 6. 2. 6 Malfunctions of Mitel DECT cordless phones

Tab. 105 Malfunctions of Mitel DECT cordless phones

Error description	Error cause / error handling
No display.	• Switch cordless phone on and test • Replace or charge battery
No radio link to radio unit; no aerial symbol.	Check coverage area (within range of a radio unit). • Check radio units in this section Cordless phone not registered with the system • Cordless phone registered
Impossible to dial.	Keypad blocked (keylock) • Unlock keypad
No dial tone.	• Check radio units in this section
Poor connection quality (echo effect).	• Turn back loudspeaker opposite (for call parties)
Cordless phone beeps approx. every 10 s during a call (or in standby) while battery indicator is flashing.	• Replace battery immediately, either after or during the call (see cordless phone user’s guide)
Call breaking up.	You are moving out of range. • Find a location with a better radio contact

Error description	Error cause / error handling
A cordless phone is called from a different system phone, but cannot be reached.	Busy tone obtained and display reads *Busy* • Cordless phone is busy Congestion tone obtained and display reads *Circuit overload*. • All radio channels busy If congestion tone is obtained after 8 seconds and display reads *No answer*. Reasons why the cordless phone could not be reached: • It is switched off • It is not within reachable radio area • No radio channels currently available • It is not registered with the system • Call diverted due to unobtainable
Cordless phone is not ringing.	• Activate tone ringing
The cordless phone cannot be configured; PIN missing (or forgotten).	• Reset PIN for user (overwrite)

6. 6. 2. 7 Malfunctions of the DECT charging bays

Tab. 106 Malfunctions of the DECT charging bay

Error description	Error cause / error handling
The cordless phone will not charge.	• Connect power supply • Check the charging contacts • Check battery and replace if necessary. About the charging process: • Battery symbol on the cordless phone is flashing (Office 135) or filling up (Office 160, Mitel 600 DECT) when the battery is being charged. • Check tone indicates correct contact.

6. 6. 2. 8 Longclicks on Mitel DECT cordless phones

In normal DECT cordless phone operation, long-clicking the following keys accesses additional functions directly.

Tab. 107 Longclicks on Mitel DECT cordless phones

Function	Office 135	Office 160	Mitel 600 DECT
In a list box: change scroll direction. Long-click "" switches to "" and vice versa	Foxkey right	Foxkey right	—
Direct access to the configuration menu	M	M	—
Switch cordless phone on/off	C, 0	0	End key
Switches over to the next radio system temporarily.	1	1	2
Indicates the radio system parameters (cordless phone IPEI and radio system PARK). With each additional call the next radio system is indicated in each case if there are other logons.	2	2	—
Indicates the cordless phone's internal diagnostics.	3	3	—
Switches to a special alarm menu of the cordless phone.	—	—	3[1)]
Indicates the data of the valid radio unit ("Show Measurement Mode", see "Planning DECT Systems" in the User's Guide).	4	4	—
Indicates the cordless phone's firmware version.	5	5	—
Jumps to the cordless phone's service menu.	—	—	5
Indicates battery charge status and the type.	6	—	—
Indicates the communication server's software version.	7	7	—
Activates "semi" key lock. See Operating Instructions for details.	8	8	—
Activates key lock. See Operating Instructions for details.	9	9	#
Switch dialling type DTMF on/off. See Operating Instructions for details.	*	*	—
Switch tone ringing on/off.	—	—	*
Jumps to the cordless phone's tone ring menu.	Loud-speaker key	Loud-speaker key	—
Menu for display contrast, display backlighting, area tone and overload tone. See Operating Instructions for details.	#	#	—
Configuration mode for hotkey. See Operating Instructions for details.	Hotkey	Hotkey	Hotkey
Switch error messages on/off (default value: Off). Messages relating to the following errors cannot be switched on/off: HS logon error, incorrect location registration, no locatable radio unit, network, system or radio unit overload.	5 + 3	5 + 3	—

1) Mitel 630 DECT only

6. 6. 2. 9 Overload code displays Office 135 / Office 160

The overload code displays on the cordless phones Office 135 and Office 160 can be activated and deactivated using the following key combination (toggle function): Long-click key 5 and then long-click key 3 (long = long-click = 2 seconds).

The overload code display is always deactivated after system initialization.

Tab. 108 DECT overload code displays Office 135

Code	Name	Error description	Error handling
05 / 06	IPEI Not Accepted	Cordless phone already registered with the system under a different number.	• Delete cordless phone registration. • Try again
10	Authentication failed	Registration error	Try again
51	DL 04 Expiry	Timer (on cordless phone) has expired	Try again
70	Timer Expired	MM timer in system has expired (during registration)	Try again
44	Failure to set up traffic bearer	Connection cannot be set up as too many cordless phones are phoning within the same range	• Try again • If still unsuccessful after several attempts, restart cordless phone and try again.
45	No Quiet Channel	No channel available, same as code 44	Same measures as for code 44
80	Reject Location Area. Not allowed. Mis-used to indicate wrong "design" version.	Wrong mode during logon.	Logon to the system < I5 • Office 135: Longclick "Home" Logon to the system > I5: • Office 135: Shortclick "Home"

6. 6. 3 Other aids

6. 6. 3. 1 System logs

During operation or in the event of a malfunction the communication server stores the current operating data in the file system in the directory */home/mivo400/logs*.

You can open, view and back up these log files on any storage device, in WebAdmin in the *System logs*(*=1w*) view.

6. 6. 3. 2 File system state

In the *File system state* (*=e3*) view you can see the thematically structured file system's memory load. In an AIN the file systems for all nodes can be viewed.

6. 6. 3. 3 File browser

With the *File browser* (*=2s*) you have access to the communication server file system and you can create new folders as well as view, import, replace or delete files in the file system.

There are the two main areas */home/mivo400/* and */ram/*. Statistical data are stored in the RAM area while all communication server folders and files are placed in the home directory.

Note:
Be extremely careful while replacing or deleting files. The absence of files can hamper or even render impossible the working of the communication server.

6. 6. 3. 4 Measuring equipment for cordless systems

The aids required for measuring out DECT systems are described under “Planning DECT Systems” in the User’s Guide.

7 Annex

This chapter informs you about the systematic designation system and provides you with an equipment overview of the communication server with cards, modules and optional components. It also provides the technical data for interfaces, communication server and system terminals as well as a table overview of the digit key assignments and function keys for the system phones. Finally here is a list of functions and products no longer supported, licence information on third-party software products, and a table summary of related documents and online help.

7.1 Systematic designation system

Tab. 109 PCB Designation

	BBBNNN.LLA.KKKKKKKKK.FF-GV
PCB type (three-digit)	
Project number (three-digit)	
Country code and sales channel	
ID	
Colour code on terminals	
Generation and version	

Tab. 110 Explanation of the PCB Designation

Part of the PCB designation	Remarks and examples
PCB type (three-digit)	LPB = Printed circuit board fitted KAB = Cable fitted PBX = Complete system SEV = Set packed EGV = Terminal packed MOV = Module/card packed
Project number (three-digit)	??? (System Mitel SMBC) 958 (System Mitel 470)
Country code and sales channel (one to three-digit, with full stops)	Two-digit country code as per ISO 3166, Sales channel (1...9) for various sales channels. Example: EXP = Export channels (not country-specific) Space = No country code

Part of the PCB designation	Remarks and examples
ID	4FXS = analogue terminal card with 4 FXS interfaces
Colour code on terminals	Colour designation in accordance with EU directive
Generation and version	Example: –3C = 3. Generation, Version C (Generation new modules: -1) Notes: • A generational change is effected following substantial changes to the functionality of a PCB. • A change of version is effected following small changes to functions or once faults have been remedied. Backward compatibility is guaranteed.

7. 2 Rating Plate and Designation Stickers

Fig. 87 Rating plate Mitel 470 communication server

Fig. 88 Designation stickers (example interface card)

7. 3 Equipment Overview

Tab. 111 Equipment Overview

Description
Mitel 470 basic system with CPU1 call manager card
3-pin network connection cable[1)]
Applications card CPU2-S
DSP module SM-DSPX1
DSP module SM-DSPX2
IP Media module EIP1-8

Description
IP Media module EIP1-32
4TAX call charge module[2)]
8TAX call charge module[2)]
16TAX call charge module[2)]
1PRI ISDN primary trunk card[3)]
1PRI-T1 ISDN primary trunk card[4)]
2PRI ISDN primary trunk card[3)]
4BRI ISDN basic trunk card/terminal interface card
8BRI ISDN basic trunk card/terminal interface card
4FXO analogue trunk card[2)]
8FXO analogue trunk card[2)]
16FXO analogue trunk card[2)]
Terminal card 8DSI
Terminal card 16DSI
Terminal card 32DSI
Terminal card 4FXS
Terminal card 8FXS
Terminal card 16FXS
Terminal card 32FXS
Fan-out panel FOP
Auxiliary power supply unit with fastening kit(APS2)
Redundant fan unit on fastening frame (RFU)
Prefabricated system cable 4 x RJ45, 6 m[3)]
Prefabricated system cable 12 x RJ45, 6 m[3)]
Prefabricated system cable 4 x RJ45, 7.62 m[4)]
Prefabricated system cable 8 x RJ45, 7.62 m[4)]
RJ45 patch cable, blue, screened, 1 m
RJ45 patch cable, blue, screened, 2 m

1) Version varies from country to country

2) The availability/release depends on the sales channel.

3) Must not be used in USA/Canada.

4) Must only be used in USA/Canada.

Tab. 112 Overview of spare parts

Description
Call manager card CPU1 (without RAM, Flash, EIM)
RAM module for call manager card CPU1
Flash module for call manager card CPU1
EIM card for call manager card CPU1
Fan with fastening screws

7. 4 Technical data

7. 4. 1 Network interfaces

The following technical data applies to the network interfaces:

Primary rate interface PRI

- E1 ISDN PRI
  - 30 B channels, 1 D channel, Bitrate 2.048 Mbit/s
  - Protocol DSS1 (public), QSIG/PSS1 (private) – used mainly in Europe
  - Protocol CAS MFC R2 – used in Brazil
  - Only on 1PRI/2PRI card
- T1 ISDN PRI
  - 23 B channels, 1 D channel, Bitrate 1.544 Mbit/s
  - Protocols: 4ESS and 5EES (AT&T), DMS100 (Nortel), National ISDN 2 (Bellcore)
  - Used in USA/Canada
  - Only on 1PRI-T1 card

Basic rate interface BRI-T

- Standard Euro ISDN interface as per CTR-3
- Configurable for point-to-point or point-to-multipoint operation
- Not usable in USA/Canada for the public network

Analogue network interfaces

- Voice path with A/D and D/A conversion (standard PCM, A-law)
- Transmission as per ES 201 168 (level country-specific)
- Signalling as per TBR 21
- Pulse or DTMF dialling, Flash signal
- Loop current detection
- Call charge receive 12 or 16 kHz (frequency and level setting country-specific)
- CLIP detection in accordance with ETS 300 778-1

7. 4. 2 Terminal interfaces

The following technical data applies to the terminal interfaces:

Digital terminal interface DSI

- Proprietary interface, two-wire
- Two system phones of the MiVoice 5300 series can be connected per interface (AD2 protocol)
- One system phone of the Dialog 4200 series can be connected per interface (DASL protocol)
- One SB-4+/SB-8 radio unit can be connected (with 8 channels the SB-8 radio units requires two DSI interfaces)
- Power supply min. 75 mA, limiting at approx. 80 mA, terminal voltage 36…48 V
- Line termination in the phone
- Transparent transmission of 2 PCM channels

Digital terminal interface BRI-S

- Standard Euro ISDN interface
- Phantom power supply min. 140 mA, limiting at approx. 170 mA, terminal voltage 36…41 V
- Up to 8 terminals can be connected
- Maximum of 2 simultaneous call connections

Analogue terminal interface FXS

- Configurable multifunctional interface for connecting analogue terminals and equipment.
- The following applies for the FXS mode *Phone / Fax*, *two-wire door* and *general bell*:
  - Voice path with A/D and D/A conversion (standard PCM, A-law)
  - Transmission as per ES 201 168 (level country-specific)
  - Constant-current loop supply approx. 25 mA (with loop resistance ≤ 1000 Ω)
  - Receive pulse or DTMF dialling
  - CLIP display on all analogue terminal interfaces simultaneously.
  - Ringing supply 40…43 V 50 Hz at load 4kΩ; no DC voltage overlay (country-specific versions also with 25 Hz)
  - No control key detection
  - No charge signalling pulses
- For more technical details and cable requirements see "Multifunctional FXS interfaces", page 151.

7. 4. 3 Communication server

Tab. 113 Dimensions and weights

	Mitel 470
Height	85 mm
Width	481 mm
Depth	380 mm
Weight (with call manager card but without mains cord, interface cards, modules and packaging)	6.71 kg

Tab. 114 Electrical isolation of interfaces

Interface	Mitel 470	
Analogue network interfaces	0.2 kV	Operating isolation
Digital network interfaces BRI		Operating isolation
Control input on FXS interface		no isolation
Control output on FXS interface		no isolation
Audio input on FXS interface		no isolation

Tab. 115 Ambient conditions

Condition	Mitel 470
Ambient temperature	5 °C to 45 °C
Relative air humidity	30 % to 80 %, non-condensating

Tab. 116 Electrical data

	Internal power supply Mitel 470	Auxiliary power supply unit (APS2)
Class of protection	1	1
Input voltage	103 V...127 V or 207 V...253 V, 48...62 Hz	100 V…240 V, 48...62 Hz
Input current	approx. 0.2 A...2.2 A (with 115 V) approx. 0.1 A...1.1 A (with 230 V)	approx. 0.2 A...4.0 A (with 115 V) approx. 0.2 A...2.0 A (with 230 V)
Resistant to voltage breaks	< 20ms	< 20ms
Power input with min. configuration	approx. 25 W	approx. 25 W
Power input with max. configuration	approx. 140 W	approx. 260 W
Undervoltage limit (system reset, data backup)	< 90 V	< 90 V

Tab. 117 Heat dissipation

	Mitel 470
Basic system with auxiliary power supply unit	approx. 140 W = 504 kJ/h
Maximally configured system	approx. 400 W = 1440 kJ/h

7. 4. 4 Dimensions of cards and modules

Tab. 118 Dimensions of cards and fan-out panels

Card	Dimensions width x height x depth [mm]
Interface cards	93 x 41 x 265
Call manager card CPU1	154 x 41 x 265
Applications card CPU2	154 x 41 x 265
Fan-out panel FOP	481 x 44 x 69

Tab. 119 Modules

Card	Dimensions length x width [mm]
DSP module	90 x 56
IP Media module	85 x 85
Charge module	83 x 60

7. 4. 5 LAN switch

10Base-TX / 100Base-TX / 1Gb-TX switch
Fully compliant with IEEE 802.3/802.3u
Auto MDI-X, Autopolarity, Autonegotiation
Flow control fully supported (half duplex: backpressure flow control, full duplex: IEEE 802.3x flow control)
Embedded SRAM for packet storage
1024-entry look-up table, direct mapping mode
QoS: 802.1p VLAN tag, DiffServ/TOS field in TCP/IP header, IP-based priority

Fig. 89 LAN switch on CPU card CPU1

100Base-TX
Fully compliant with IEEE 802.3/802.3u
Embedded SRAM for packet storage
1024-entry look-up table, direct mapping mode
QoS: 802.1p VLAN tag, DiffServ/TOS field in TCP/IP header, IP-based priority

Fig. 90 LAN switch on the backplane

7. 4. 6 Digital and IP system phones

Tab. 120 Digital and IP system phones

	MiVoice 5360 / 5360 IP, MiVoice 5361 / 5361 IP, MiVoice 5370 / 5370 IP, MiVoice 5380 / 5380 IP
Ambient temperature in operation	0 °C to 40 °C
Relative humidity in operation	30 % to 80 %

	MiVoice 5360 / 5360 IP, MiVoice 5361 / 5361 IP, MiVoice 5370 / 5370 IP, MiVoice 5380 / 5380 IP
Admissible storage temperature	-25 °C to 45 °C
Power consumption, digital system phones	see table "Average power requirements of terminals", page 93 and table "Maximum power requirements of the system phones on the DSI bus", page 139
Power consumption, IP system phones	see System Manual for "Mitel Advanced Intelligent Network (AIN) and IP system phones"

Tab. 121 Dimensions and weights, digital and IP system phones

Terminals	Height (Type of mounting)	Width	Depth (Type of mounting)	Weight
MiVoice 5360, MiVoice 5360 IP, MiVoice 5361, MiVoice 5361 IP	115 mm (desktop 25 °) 151 mm (desktop 45 °) 199 mm (wall)	262 mm	198 mm (desktop 25 °) 166 mm (desktop 45 °) 90 mm (wall)	approx. 850g
MiVoice 5370, MiVoice 5370 IP	115 mm (desktop 25 °) 151 mm (desktop 45 °) 199 mm (wall)	262 mm	198 mm (desktop 25 °) 166 mm (desktop 45 °) 90 mm (wall)	approx. 875 g
MiVoice 5380, MiVoice 5380 IP	115 mm (desktop 25 °) 151 mm (desktop 45 °) 199 mm (wall)	262 mm	198 mm (desktop 25 °) 166 mm (desktop 45 °) 90 mm (wall)	approx. 935 g
Expansion key module MiVoice M530	115 mm (desktop 25 °) 151 mm (desktop 45 °) 199 mm (wall)	95 mm	198 mm (desktop 25 °) 166 mm (desktop 45 °) 90 mm (wall)	approx. 180 g
Expansion key module MiVoice M535	115 mm (desktop 25 °) 151 mm (desktop 45 °) 199 mm (wall)	128 mm	198 mm (desktop 25 °) 166 mm (desktop 45 °) 90 mm (wall)	approx. 325g

7. 4. 7 Mitel DECT radio units

GAP functionality

The following table contains the network features as defined in the GAP standard. For each feature a separate column indicates whether it is supported by communication servers of the MiVoice Office 400 family or Mitel DECT cordless phones.

Tab. 122 **Features supported as per GAP standard**

No.	Feature	PP	In Mitel DECT cordless phones	FP	In MiVoice Office 400
1	Outgoing call	M	✓	M	✓
2	Off hook	M	✓	M	✓
3	On hook (full release)	M	✓	M	✓
4	Dialled digits (basic)	M	✓	M	✓
5	Register recall	M	✓	O	✓
6	Go to DTMF signalling (defined tone length)	M	✓	O	✓
7	Pause (dialling pause)	M	✓	O	—
8	Incoming call	M	✓	M	✓
9	Authentication of PP	M	✓	O	✓
10	Authentication of user	M	✓	O	—
11	Location registration	M	✓	O	✓
12	On air key allocation	M	✓	O	✓
13	Identification of PP	M	✓	O	—
14	Service class indication / assignment	M	✓	O	—
15	Alerting	M	✓	M	✓
16	ZAP	M	✓	O	—
17	Encryption activation FP initiated	M	✓	O	—
18	Subscription registration procedure on-air	M	✓	M	✓
19	Link control	M	✓	M	✓
20	Terminate access rights FP initiated	M	✓	O	✓
21	Partial release	O	✓	O	✓
22	Go to DTMF (infinite tone length)	O	—	O	—
23	Go to Pulse	O	—	O	—
24	Signalling of display characters	O	✓	O	—
25	Display control characters	O	—	O	—
26	Authentication of FP	O	✓	O	✓
27	Encryption activation PP initiated	O	—	O	—
28	Encryption deactivation FP initiated	O	—	O	—
29	Encryption deactivation PP initiated	O	—	O	—
30	Calling Line Identification Presentation (CLIP)	O	✓	O	✓
31	Internal Call	O	✓	O	—
32	Service Call	O	—	O	—

PP: Portable Part
FP: Fixed Part
M: Mandatory (this feature must be supported by GAP compliant equipment)
O: optional
—: The Mitel DECT cordless phones and MiVoice Office 400 communication servers do not support the feature.

Technical data

Tab. 123 Mitel DECT radio units

Duplex method	Time-division multiplex, 10 ms frame length
Frequency range	1880 MHz to 1900 MHz
Frequency bands (carrier)	10
Channel spacing (carrier distance)	1,728 MHz
Transmission rate	1152 kbit/s
Duplex channels per carrier SB-4+ / SB-8	6 / 12
Number of channels (duplex channels) SB-4+ / SB-8	60 / 120
Modulation	GFSK
Data transfer rate	32 kbit/s
Voice encoding	ADPCM
Transmit power	250 mW peak value 10 mW, average power per channel
Range	30 to 250 m
Max. line length to radio unit - power supply via DSI bus (0.5mm) - with power supply unit (9–15 VDC, 400 mA)	 1200 m 1200 m
Ambient temperature, radio unit in operation	-10 °C to 55 °C
Admissible storage temperature	-25 °C to 55 °C
Relative humidity in operation	30 % to 80 %
IP class of protection	IP 30
Dimensions: Radio unit W x H x D:	165 x 170 x 70 mm
Weight: Radio unit	320 g
Local power supply to radio unit (optional)	Plug-in power supply unit

7. 5 Operation of digital system phones

7. 5. 1 Digit key assignment of system phones

Digit key assignment depends on the system phones series and the language set for the communication server.

The following Latin script assignment for the digit keys applies to the system phones / MiVoice 5360 / 5360 IP, MiVoice 5361 / 5361 IP, MiVoice 5370 / 5370 IP, Office 135/135pro and all models of Office 160 for all communication server languages with the exception of Greek:

Tab. 124 Latin-script digit key assignment

Key	Characters	Key	Characters
1	- . ? 1 ! , : ; ' " ¿ ¡ - . ? 1 ! , : ; ' " ¿ ¡	2 ABC	A B C 2 Ä Æ Å Ç a b c 2 ä æ å à ç
3 DEF	D E F 3 É d e f 3 é è ê	4 GHI	G H I 4 g h i 4 ì
5 JKL	J K L 5 j k l 5	6 MNO	M N O 6 Ñ Ö Ø m n o 6 ñ ö ø ò
7 PQRS	P Q R S 7 p q r s 7 ß	8 TUV	T U V 8 Ü t u v 8 ü ù
9 WXYZ	W X Y Z 9 w x y z 9	0	+ 0 + 0
*	* / ( ) < = > % £ $ ¤ ¥ ¤ @ & § * / ( ) < = > % £ $ ¤ ¥ ¤ @ & §	#	Space # Space #

Notes:
- The MiVoice 5360 phones do not have a graphics-compatible display and therefore cannot display all the characters featured (see also the corresponding user guide).
- On the Office 160 cordless system phone the space character is stored under digit 0 and the special characters are stored under the #-key instead of the *-key.

7. 5. 2 Alpha keyboardMiVoice 5380 / 5380 IP

The integrated alphanumerical keyboard on the MiVoice 5380 / 5380 IP is available in a QWERTY and AZERTY version. The special characters can be called up using the "Ctrl" key and the "Shift" key.

Tab. 125 Integrated alphanumerical keyboard MiVoice 5380 / 5380 IP

Key	<Key>	Shift + <key>	Ctrl + <key>	Ctrl + Shift + <key>
A	a	A	ä á à â ã å æ	Ä á à â Ã Å Æ
B	b	B		
C	c	C	ç	Ç
D	d	D		
E	e	E	é è ê ë	É è ê Ë
F	f	F		
G	g	G		
H	h	H		
I	i	I	ï í ì î	ï í ì î

Key	<Key>	Shift + <key>	Ctrl + <key>	Ctrl + Shift + <key>
j	j	j		
K	k	K		
L	l	L		
M	m	M		
N	n	N	ñ	Ñ
O	o	O	ö ó ò ô õ ø	Ö ó ò ô Õ Ø
P	p	P		
Q	q	Q		
R	r	R		
S	s	S	ß	
T	t	T		
U	u	U	ü ú ù û	Ü ú ù û
V	v	V		
W	w	W		
X	x	X		
Y	y	Y	ÿ	
Z	z	Z		
@	@	@		
+	+	+	- . ? ! , : ; . " / \ ( ) = < > % £ $ õ ¥ ª & § ¿ ¡	

7. 5. 3 Function commands (macros)

Function commands are used mainly for automatically activating/ deactivating features using the function keys of the system phones. The following function commands are available:

Tab. 126 Function commands for system phones

Function command	Meaning
"A"	Seize line with maximum priority[1)]
"I"	Seize line
"H"	Seize line in hands-free mode[2)]
"X"	Disconnect
"P"	Pause 1 second before next action
"Lxx"	Seize line xx (line keys)[1)]
"N"	Enter call number keyed in during call preparation
"."	Control keys function
"Z"	Activate / deactivate DTMF mode (tone dialling)
"R"	Use call number last dialled
"Y"	End call and reseize line

1) Available only with the key telephones.

2) Available for Mitel 600 DECT only.

The function commands can be stored directly on the system phones via Self Service Portal or on the function keys via WebAdmin.

7. 6 Functions and terminals no longer supported

The MiVoice Office 400 series continues to support the terminals and functions of the Aastra IntelliGate series. Exceptions include the following terminals and functions:

-
- IP system phones Office 35IP, Office 70IP-b
- Cordless system phones Office 100, Office 130/130pro, Office 150, Office 150EEx, Office 155pro/155ATEX
- The Aastra 6751i phone is no longer supported as an Mitel SIP phone.
- IP system softphone Office 1600/1600IP
- DECT radio unit SB-4
- Pocket Adapter V.24
- X.25 in the D channel
- Ascotel® Mobility Interface (AMI) and DCT terminals
- Universal Terminal Interface (UTI)
- AMS Hotel manager and Hospitality Mode V1.0 (hotel functions)
- Operator application Office 1560/1560IP
- Aastra Management Suite (AMS) is replaced by the web-based configuration tool WebAdmin, the remote management SRM (Secure IP Remote Management) and the application System Search.
- The external remote control (ERC) cannot be set up with WebAdmin. ERC is replaced by the possibility, to integrate mobile phones and other external phones into the system (Mobile or External Phone Extension).
- Only language package downloading is available for Virtual Appliance in System Search, Emergency Upload and the display of Virtual Appliance communication servers is not available.
- Mitel BluStar 8000i is not supported by the Virtual Appliance communication server.
- The CPU2 application card is no longer supported (only CPU2-S).
- The Telephony Web Portal (TWP) application is replaced with Mitel MiCollab Audio, Web and Video Conferencing.

7. 7 Licensing information of third-party software products

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Copyright and License Information

7. 8 Documents and online help systems with further information

Product	Document
Products of the MiVoice Office 400 family	System Manual Mitel 415/430
	System Manual Mitel SMBC
	System Manual Virtual Appliance
	System Manual System Functions and Features
	SIP Access User's Guide (English)
	MiVoice Office 400 feature overview
Applications card CPU2-S	User's Guide MiVoice Office 400 fax service (German and English only)
	Installation Guide Applications Card CPU2-S
Applications	System Manual Mitel Alarm Server
	Mitel Alarm Server User's Guide
	Installation Instructions Mitel OpenCount for MiVoice Office 400
	Configuration Guide Mitel OpenCount for MiVoice Office 400
	Installation and Administration Guide "Mitel Standard Linux"
	Solutions Guide "Virtual Appliance Deployment"
	Mitel SIP Teleworker via MBG on MiVoice Office 400
SMBC Manager	Online Help
WebAdmin	Online Help
	Configuration assistant
	Setup wizard
Self Service Portal (SSP)	Online Help
Project planning application Mitel CPQ	Online Help
DECT	Planning DECT systems User's Guide
Mitel SIP-DECT	User's Guide for Mitel 600 SIP-DECT on MiVoice Office 400
Basic/Enterprise voice mail system	User's Guide for MiVoice Office 400 voice mail system
	System Manual System Functions and Features
OIP	System Manual Mitel Open Interfaces Platform
	Online Help
	User's Guide Mitel OfficeSuite
	User's Guide for First Party TAPI Service Provider
Networking	System Manual for Mitel Advanced Intelligent Network (AIN) and IP system phones
	Private networking system manual
Mitel SIP phones on MiVoice Office 400	Mitel 6730/31/53 SIP, Mitel 6735/37/55/57 SIP, Mitel 6739 SIP, Mitel 6863/65 SIP, Mitel 6867/69 SIP, Mitel 6873 SIP, Mitel 6920 SIP/Mitel 6930 SIP, Mitel 6940 SIP user's guide

Product	Document
Mitel SIP phones (platform-independent)	User's guide, short user's guide, installation instructions, administration instructions
IP system phones	Quick User's Guide MiVoice 5360 IP / MiVoice 5361 IP / MiVoice 5370 IP /MiVoice 5380 IP Operating Instructions for MiVoice 5360 IP / MiVoice 5361 IP / MiVoice 5370 IP / MiVoice 5380 IP / MiVoice 2380 IP
Digital system phones	Quick User's Guide Office 135/135pro / Office 160pro/Safeguard/ATEX / MiVoice 5360 / MiVoice 5361 / MiVoice 5370 / MiVoice 5380 / Mitel 610 DECT / Mitel 612 DECT / Mitel 620 DECT / Mitel 622 DECT / Mitel 630 DECT / Mitel 632 DECT / Mitel 650 DECT User's Guide Office 135/135pro / Office 160pro/Safeguard/ATEX / MiVoice 5360 / MiVoice 5361/ MiVoice 5370/ MiVoice 5380 / MiVoice 5380 / Mitel 610 DECT / Mitel 612 DECT / Mitel 620 DECT / Mitel 622 DECT / Mitel 630 DECT / Mitel 632 DECT / Mitel 650 DECT / Dialog 4220 / Dialog 4222 / Dialog 4223
Analogue phones	Mitel 6710 Analogue / Mitel 6730 Analogue user's guide
PC operator console	User's Guide MiVoice 1560 PC Operator Online Help

Most of the documents are accessible at http://www.mitel.com/docfinder. Many documents in the above table are summarised per language and software release in documentation sets, and can be downloaded as a .zip file. Note: Documentation sets are very large (~500 MB). The download can take some time, depending on the connection.

More documents are available on the internet:

- Environmental information for communication server and system phones
- Declarations of conformity for communication server and system phones
- Labels for system phones and expansion key modules
- Safety instructions for system phones
- Application Notes
- Product information
- Leaflets
- Brochures
- Data sheets

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