With a passion for secure communication

SIP stations by Commend are designed to always connect you to the right contact person – whether at the car park gate or entrance door, at the bus stop or information/emergency call terminal.

Reliably and in perfect speech quality.

Compatible with SIP PBX Server

Digium | Cisco | Avaya | Alcatel | Mitel | Siemens | 3CX | Starface | Aastra | Kamailio | FreeSWITCH | ELMEG | Unify | AVM | Innovaphone | and many more ...

Serverless Communication

SIP stations and SIP capable telephones interface seamlessly into an intelligent communication network without the need for a server

- Perfect HD Voice speech quality
- Action sequences
- Output controlling
- Input monitoring
- Automatic voice message playback
- Location identification messages
- Wide range of functions
- Easy configuration
- Low energy costs
- Security through redundancy

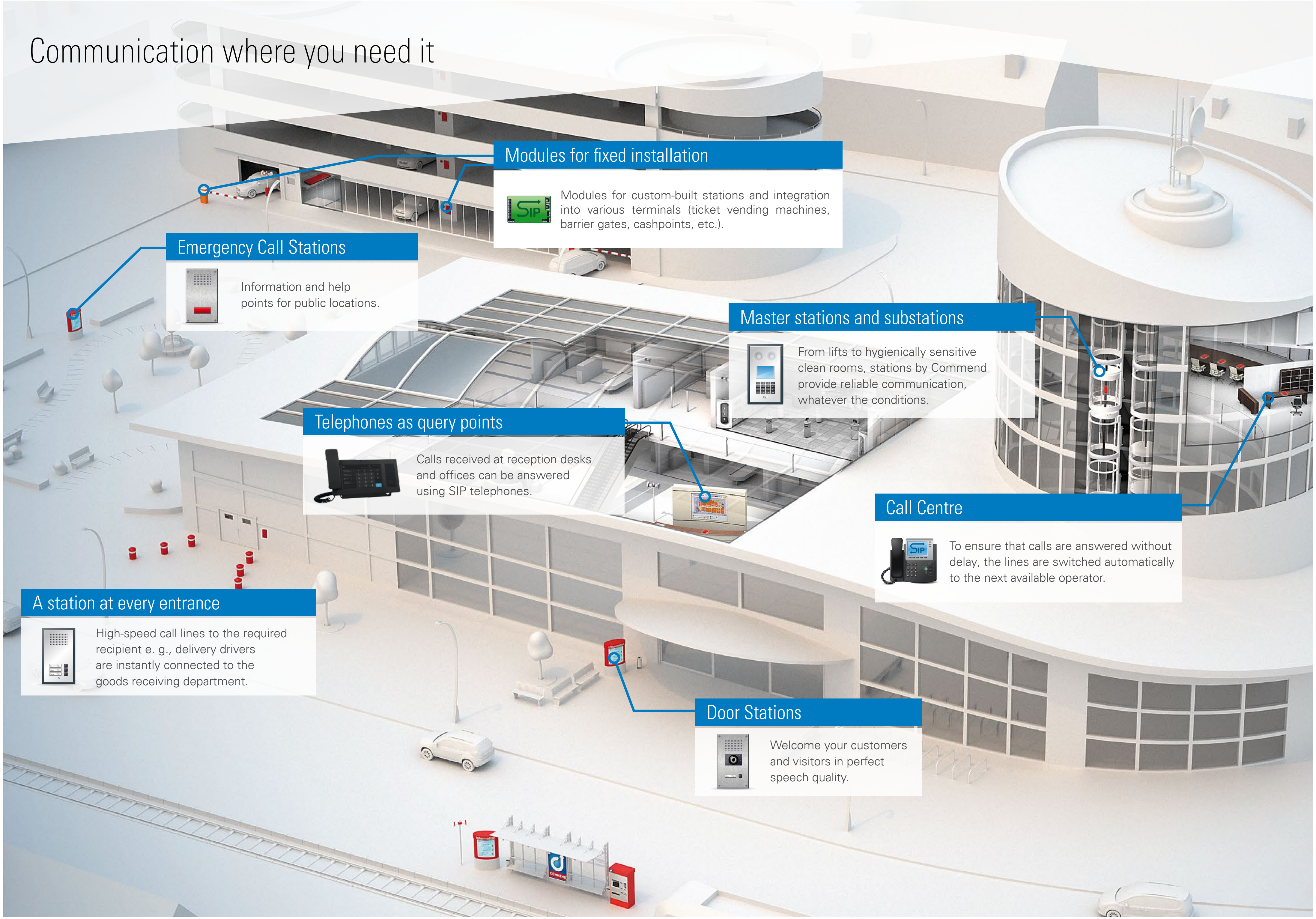
Vehicle entry and exit gates

Entrances and airlocks

Ticket vending machines

Information and emergency terminals

Communication where you need it



Modules for fixed installation

Modules for custom-built stations and integration into various terminals (ticket vending machines, barrier gates, cashpoints, etc.).

Emergency Call Stations

Information and help points for public locations.

Master stations and substations

From lifts to hygienically sensitive clean rooms, stations by Commend provide reliable communication, whatever the conditions.

Telephones as query points

Calls received at reception desks and offices can be answered using SIP telephones.

Call Centre

To ensure that calls are answered without delay, the lines are switched automatically to the next available operator.

A station at every entrance

High-speed call lines to the required recipient e. g., delivery drivers are instantly connected to the goods receiving department.

Door Stations

Welcome your customers and visitors in perfect speech quality.

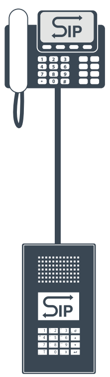
Serverless communication

Commend stations come with enough
built-in intelligence to make a server unnecessary.

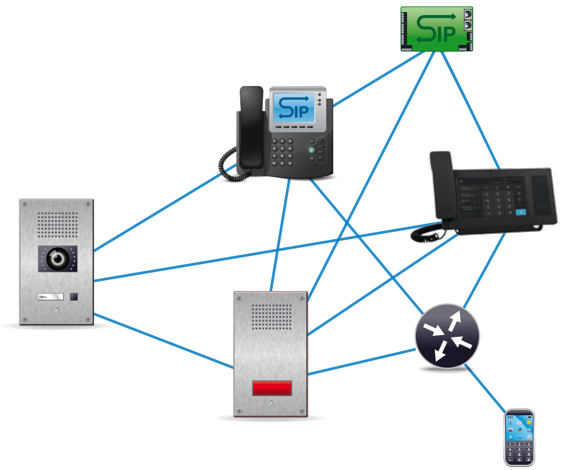
SIP stations and SIP capable telephones interface seamlessly into an intelligent, independent communication network without the need for a controlling server.

- Cost-efficient and ideal for small-scale applications
- Calls can be made in both directions
- Calls can be forwarded to the telephone network via a gateway
- Quick and easy to set up and start up
- Re-dialling, e.g., for opening doors

Direct connections
without the need for a server



Intelligent communication networks
without the need for a server



More than just communication

Relays and attendant contacts enable powerful control functions

Stations come with the ability to remote-control relays.

- Doors, shutters, gates and barriers open effortlessly at the touch of a button (desktop or mobile telephone) or by remote control via a third-party system (http request)
- Easy control of signal lamps and other subsections

Attendant contacts for additional indication of operating states such as Error, Ringing, Active Call, etc. (e. g., automatic activation of flashing light signal to indicate incoming calls).

Server support –
one common language

In the most common case where a server-based SIP PBX telephone system is already in place, our stations integrate seamlessly with the server to ensure maximum mutual benefit.

Especially in larger-scale systems, both components enhance each other by offering functions that would normally require a SIP server.

Quickly assign calls and reduce waiting times

In serverless communication scenarios the next free query point is found by calling each one using an Action Sequence. Server integration, on the other hand, allows for incoming calls to be allocated instantly and automatically to the next available operator (e. g., at a call centre). This way, waiting times for callers are reduced to an absolute minimum.

Easy maintenance of multiple stations

With the help of a provisioning mechanism, configurations and firmware for functional updates can be deployed to all stations within the network simultaneously to save time and effort.

More security through Server Redundancy

When calling for help at an Emergency Call Station, people must be able to rely on the call getting through under any circumstances. This is where server redundancy comes in. If a server happens to fail, another one takes over seamlessly to ensure that the call is put through.

Ready for your system

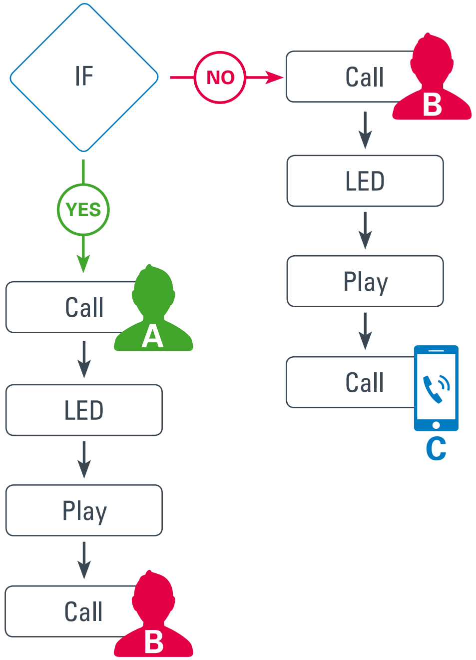
Full compliance with the SIP standard and its basic feature set makes our stations compatible with virtually any SIP PBX system available today. Compatibility is tested extensively both with current and new systems.

Stations with a server's intelligence

Action Sequences

The stations can be programmed with Action Sequences, which allow the station to perform multiple functions and processes automatically in a specific sequence – no server required!

Call, Relay, IF-THEN, PLAY and LED components can be combined into customisable sequences. Multiple sequences can also be linked together to be executed in the specified order.



If a call is made on a weekday (Monday to Friday) between 8 am and 5 pm, ...

... then dial number **A**.

... the LED lights up to indicate that the connections is being made..

... a voice message is played back.

If Subscriber **A** fails to respond after a pre-defined time, the system automatically dials number **B**.

... otherwise, dial number **B**.

... the LED lights up to indicate that the connections is being made..

... a voice message is played back.

If Subscriber **B** fails to respond after a pre-defined time, the system automatically dials the number of mobile phone **C**.

Examples

- If the call is not answered within a pre-defined time, it is relayed automatically to the next receiving station in the chain
- Incoming calls can be forwarded to any number of receiving stations. Optionally, the call (usually an emergency call) can also be forwarded up to 22 receiving stations simultaneously without a server
- Depending on the type of call, various modes can be activated automatically – e. g., flashing blue signal light for emergency calls, normal signal light for information calls, etc.
- Scheduling of actions for specific weekdays or times
- Playback of pre-recorded voice messages
- Control of outputs within an action sequence
- Triggering of specific action sequences by an input signal, or waiting for an input signal (e. g., air lock)

... and many other possible combinations

7kHz

Optimum speech intelligibility

A loud, clear and beautifully crisp voice signal ensures natural, face-to-face style communication with visitors and customers – even in challenging of situations

- **Suppression of interfering background sounds** such as traffic noise
- Easy to hear, thanks to **higher volume capacity** than standard SIP stations
- OpenDuplex® for simultaneous speaking and listening at high volume levels
- Switched Duplex for situations with extreme ambient noise (e. g., tunnels)
- HD Voice speech quality with 7 kHz audio bandwidth

Always there for you

Automated voice messages for welcoming visitors or reassuring emergency callers

Pressing the call button at an entrance or emergency call station triggers the playback of a customised voice message, reassuring the caller that someone will be available shortly to assist them.

24/7

Always at your service, thanks to redundancy

Functionality for more security

- Stations can be logged in at up to three servers simultaneously
- Calls are transmitted via the active server
- In case none of the servers can be reached, the system can try to establish a serverless connection if necessary – e.g., by calling all stations on the network

Electricity costs as low as €2.60 a year

A nice contribution to the environment

When it comes to low power consumption, Commend's SIP stations are second to none.

- Approx. 1.5 Watts in standby mode, and only 2 Watts in call mode, depending on the volume level
- Power can be supplied via PoE or an external power adapter

Wide range of functions

- Telephone directory and web call
- Connection ports for external amplifier and loudspeakers
- Connection ports for add-on modules (loudspeaker, direct dialling buttons, handset)
- SNMP for station monitoring
- http support for network-based control of stations

Where is the call coming from?

Location identification messages provide information for users at the control desk or query point

An optional **location identification message** (e. g., "Emergency Call Station at Subway Station West Park") can be defined for each station individually. The identification message is played back automatically when the operator at the control desk or query point takes the call. This way, the operator knows immediately where the call is coming from without having to ask. This is particularly important if there is no visualisation system installed at the Control Desk or Query Point, or if the call is relayed to a mobile phone.

Loudspeaker-Microphone Monitoring

Functionality that enhances security

This causes the SIP Station to emit an unnoticeable audio test signal through the loudspeaker, which is picked up and analysed by the microphone. If the test signal does not arrive in the required quality (e. g., due to chewing gum blocking the microphone), the station will notify the receiving station accordingly. This ensures constant availability without the need for regular manual inspections, which goes a long way towards saving costs.

Configuration made easy

The stations are specifically designed for easy, convenient configuration over the special Web Interface. A few clicks is all it takes to perform an update and even set up complex action sequences. For large-scale installations, the provisioning function helps to deploy configuration settings automatically and conveniently to thousands of connected stations at once.

Simply compatible

SIP Stations integrate seamlessly into existing Commend security and communication systems as needed. This allows for adding features such as announcements, audio recording, interfacing with external systems (e. g., visualisation), and many more.

SIP PBX Kompatibilität

Embracing the SIP world

Commend SIP stations can be used with a wide range of SIP PBX systems.

We are constantly performing comprehensive tests with SIP servers of different manufacturers. The list keeps growing longer to ensure uncompromising compatibility.

Digium Asterisk	Cisco Cisco Call Manager Cisco Unified Communication Manager	AVM Fritz!Box
Avaya Aura	Alcatel OmniPCX Enterprise	Unify OpenScape
Mitel	Siemens Hipath 4000 Hipath 3000 + HG 1500	Innovaphone Virtual Appliance IPVA
3CX 3CX for Windows	Starface Starface free	FreeSWITCH
Aastra MX-ONE	Kamailio Kamailio (OpenSER)	ELMEG elmeg ICT880






SIP	Vandal resistant SIP door station	Vandal resistant SIP master station	Vandal resistant SIP door station		Vandal resistant emergency call station		Disabled-friendly vandal resistant SIP station – DDA/ADA compliant	Vandal resistant SIP stations for calls and emergency calls	
Article	**SIP-WS 201V CA**	**SIP-WS 800V**	**SIP-WS 201V**	**SIP-WS 203V**	**SIP-WS 211V**	**SIP-WS 212V**	**SIP-WS 211V DA**	**EF 962H**	**EF 962HM**
Keypad	1 call button	Full keypad	1 call button	3 call buttons	1 emergency call button	1 emergency and 1 call button	1 call button	1 call button	1 emergency and 1 call button
Back-lit keypad	•	•	•		•		•	–	
Display	–	LCD graphic display 128 x 64 pixels	–		–		–	–	
Camera	Axis colour video camera, video streams H.264 (MPEG-4 Part 10/ AVC) and M-JPEG, max. resolution 1280 x 720 pixels (720p)	–	–		–		–	–	
Special features	Integration in 3rd-party Video Management Systems – VMS	–	–		–		Integrated IEC 60118-4 compliant induction loop system, LED pictograms	LED pictograms	
Front panel	Vandal resistant design, 3 mm V-2A steel, poke protected and fitted with special security screws								
IP rating	IP 65	IP 65	IP 65		IP 65		IP 65	IP 54	
IK rating	IK 09	IK 07	IK 09		IK 09		IK 07	–	
Inputs	3 x inputs for floating contacts							2 x inputs for floating contacts	
Outputs	2 x SPDT relay outputs							2 x SPDT relay outputs	
Line output	For connection of loudspeaker module							–	
Microphone	Omnidirectional electret microphone for max. 7 m (23 ft) speaking distance								
Loudspeaker	2 x 8 Ω loudspeakers							1 x 8 Ω loudspeaker	
Audio Quality	HD Voice 7 kHz, OpenDuplex®, Noise Cancelling, very high volume thanks to integrated class „D" amplifier								
IP protocol	IPv6 ready, IPv4, TCP, UDP, HTTP, RTP, RTCP, DHCP, SNMPv2, STUN								
Codecs	G.722, G.711 a-Law, G.711 µ-Law								
Cabling	min. Cat. 5								
Ethernet	2 x 10/100 MBit/s (full/half Duplex) Auto MDIX							1 x 10/100 MBit/s (full/half Duplex)	
Power supply	PoE	24 VDC ± 2 V, 500 mA or PoE					24 VDC ± 2 V; 500 mA	PoE	
PoE – Power over Ethernet	Standard IEEE 802.3af; Power consumption of the terminal device – Class 0 (0.44 W to 12.95 W)						–	Standard IEEE 802.3af	
Operating temperature range	–25° C bis +50° C	–20° C to +70° C						–20° C to +70° C	
Additional installation material	Flush-Mount Kit WSFB 50V; Flush-Mount Kit WSFB 50V SS FL; Surface-Mount Kit WSSH 50V; rain protection roof WSRR 50V							Basic housing GUEF 962 required for surface and flush mounting, surface-mount boxes EF 62, EF 62W	
Expansion options	Direct dialling button modules WSDD 59V, WSDD 53V, loudspeaker modules WSLM 56V, WSLM 52V							–	
Dimensions	Flush mounted with WSFB 50V – W 164 x H 279 x D 14 mm; Flush mounted with WSSH 50V SS FL – W 164 x H 279 x D 0 mm; Surface mounted WSFB 50V – W 164 x H 279 x D 50 mm							Flush m. W 110 x H 151 mm Surface m. W 110 x H 151 x D 55 – 84	





SIP	SIP master station	SIP door station		SIP station for medical environments and clean rooms	SIP station for medical environments and clean rooms	Compact control desk station with touchscreen	
Article	**SIP-WS 800P**	**SIP-WS 201P**	**SIP-WS 203P**	**SIP-WS 800F**	**SIP-WS 800F D MD**	**EE 980 – Duetto**	**EE 980 CM – Duetto**
Keypad	Full keypad	1 call button	3 call buttons	Full keypad	Full keypad	Touchscreen	
Back-lit keypad	•	•		•	•	–	
Display	LCD graphic display 128 x 64 pixels	–		LCD graphic display 128 x 64 pixels	LCD graphic display 128 x 64 pixels	7" IPS touch display with 800 x 480 pixels	
Camera	–	–		–	–	–	HD ready video camera 3,1 Mega-pixel
Special features	–	–		–	Anti-bacterial membrane covered surface especially for clean rooms	SIP video, customisable user interface, Gesture Control, Ambient light sensor	
Front panel	Polycarbonate			Polycarbonate, closed-sealed membrane surface, detergent and disinfectant resistant		Polycarbonate	
IP rating	IP 65	IP 65		IP 65	IP 65	IP 20	IP 20
IK rating	–	–		–	–	–	–
Inputs	3 x inputs for floating contacts					2 digital inputs for floating contacts	
Outputs	2 x SPDT relay outputs					2 digital outputs for Open Drain	
Line output	For connection of loudspeaker module					Line-in/Line-Out switch-able via headset jack	
Microphone	Omnidirectional electret microphone for max. 7 m (23 ft) speaking distance					4 microphones with directivity	
Loudspeaker	2 x 8 Ω loudspeakers					1 x 8 Ω loudspeaker	
Audio Quality	HD Voice 7 kHz, OpenDuplex®, Noise Cancelling, very high volume thanks to integrated class „D" amplifier						
IP protocol	IPv6 ready, IPv4, TCP, UDP, HTTP, RTP, RTCP, DHCP, SNMPv2, STUN					IPv6 ready, IPv4, TCP, UDP, HTTP, RTP, RTCP, DHCP, RTSP, SIP, STUN	
Codecs	G.722, G.711 a-Law, G.711 µ-Law						
Cabling	min. Cat. 5						
Ethernet	2 x 10/100 MBit/s (full/half Duplex) Auto MDIX					1 x 10/100 MBit/s (full/half Duplex) Auto MDIX	
Power supply	24 VDC ± 2 V, 500 mA or PoE					PoE or external power supply 24 VDC	
PoE – Power over Ethernet	Standard IEEE 802.3af; Power consumption of the terminal device – Class 0 (0.44 W to12.95 W)					Standard IEEE 802.3af; Power consumption of the terminal device – Class 3 (6.49 to 12.95 W)	
Operating temperature range	–20° C to +70° C			–20° C to +60° C		0° C to + 50° C	
Additional installation material	Flush-Mount Kit WSFB 50P, Surface-Mount Kit WSSH 50P, Desktop Kit WSDK 50P					Desktop Kit EE DK9; Wall-Mount Frame EE SH9	
Expansion options	Loudspeaker modules WSLM 56P, WSLM 52P, handset module WSHS 50P			Loudspeaker modules WSLM 56F, WSLM 52F, handset module WSHS 50P		Handset EE HS9, Headset HS1	
Dimensions	Flush mounted with WSFB 50P – W 165 x H 280 x D 13 mm; Surface mounted with WSSH 50P – W 165 x H 280 x D 51 mm					Desktop Kit – W 270 x H 142 x D 70 mm; Wall-Mount Frame – W 270 x 138 mm x D 38 mm	

SIP	SIP modules for assembling customer-specific stations and installing into various terminals						SIP loudspeaker for Public Address with direct IP network connection		
Article	**SIP-ET 908A**	**SIP-ET 908A-1**	**SIP-ET 908MI**	**SIP-ET 908MI1**	**ET 962H**	**ET 970H**	**AFLS 10H HG**	**AFLS 10H PW**	**AFLS 10H CW**
Keypad	Ready for connection of a keypad with 18 keys or three single buttons				Ready for connection of three single buttons				
Inputs	3 x inputs for floating contacts				2 x inputs for floating contacts				
Outputs	2 x SPDT relay outputs				2 relay outputs (1x make contact, 1x break contact – 1 of them as change over contact)				
Special features	2 horizontally installed RJ 45 ports	2 vertically installed RJ 45 ports	2 horizontally installed RJ 45 ports	2 vertically installed RJ 45 ports	1 RJ 45 port		Horn loudspeaker	Projector loudspeaker	Ceiling loud-speaker
IP rating	–				–		IP 66	IP 54	IP 54
Amplifier	2.5 W class „D" amplifier				10 W class „D" amplifier				
Integrated loutspeaker	–				1 x 8 Ω loudspeaker	–	–		
Sound pressure	–				85 dB/1 W/1 m	–	118 dB/1 m	101 dB/1 m	105 dB/1 m
Connection for ex-ternal loudspeaker	4 – 50 Ω				8 – 50 Ω	4 – 50 Ω	–		
Microphone	Microphone input for electret microphone or dynamic microphone		Microphone input for electret microphone or dynamic microphone, build-in microphone MIC 480 included		Integrated electret microphone	Integrated electret micro-phone and build-in microphone MIC 480 included	Integrated electret microphone MIC 480		Microphone input for build-in microphone MIC 480
LED	Possibility for connection of a multi-functional RGB LED				Light pipe and plexiglass LED cover, optionally as light guide for the multi-functional RGB LED		Possibility for connection of a multi-functional RGB LED		
Audio Quality	HD Voice 7 kHz, OpenDuplex®, Noise Cancelling, very high volume thanks to integrated class „D" amplifier								
IP protocol	IPv6 ready, IPv4, TCP, UDP, HTTP, RTP, RTCP, DHCP, SNMPv2, STUN								
Codecs	G.722, G.711 a-Law, G.711 μ-Law								
Cabling	min. Cat. 5								
Ethernet	2 x 10/100 MBit/s (full/half Duplex) Auto MDIX				1 x 10/100 MBit/s (full/half Duplex) Auto MDIX				
Power supply	12 – 24 VAC or 15 – 35 VDC, 500 mA or PoE				PoE				
PoE	Standard IEEE 802.3af; Power consumption of the terminal device – Class 0 (0.44 W to 12.95 W)								
Operating temperature range	–40° C to +70° C				–40° C to +70° C		–20° C to +70° C		
Dimensions	W 65 x H 130 x D 18 mm	W 65 x H 130 x D 21 mm	W 65 x H 130 x D 18 mm	W 65 x H 130 x D 21 mm	W 87.5 x H 109 x D 45 mm		W 180 x H 120 x D 230 mm	ø 145 mm x D 210 mm	ø 167 mm x D 112 mm

The design and/or specifications of products may be subject to change for improvement without prior notice. Errors excepted.

Accessories Series WS – Vandal resistant


WSDD 59V
Direct dialling button module


WSLM 56V
Loudspeaker module


WSDD 53V
Direct dialling button module


WSLM 52V
Loudspeaker module


WSSH 50V
Surface-Mount Kit


WSFB 50V
Flush-Mount Kit


WSFB 50V SS FL
Flush-Mount Kit


WSRR 50V
Rain protection roof

Accessories Series WS – Polycarbonat


WSLM 56F
Loudspeaker module


WSLM 56P
Loudspeaker module


WSLM 52F
Loudspeaker module


WSLM 52P
Loudspeaker module


WSSH 50P
Surface-Mount Kit


WSFB 50P
Flush-Mount Kit


WSDK 50P
Desktop Kit


WSHS 50P
Handset module

Accessories Series EF


GUEF 962
Basic housing for flush and surface mounting


EF 62
Surface-Mount Box

Accessories Duetto / EE 980


EE HS9
Handset


EE DK9
Desktop Kit


EE SH9
Wall-mounting frame


HS 1
Headset

A strong worldwide network

COMMEND is represented the world over by local Commend Partners and helps to improve security and communication with tailored solutions.

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