SIP-WS 800P

Resistant SIP wallmount station with LCD graphic display

Open-Duplex®

High Volume

ONVIF VMS integration

HD Voice 7 kHz Audio

Weather-proof IP66

The SIP solution for entrances and gateways

The OpenDuplex® capable SIP station SIP-WS 800P has been developed and designed for areas with high requirements on safety.

The station is connected directly with the Ethernet (LAN/WAN) and in this manner it is connected to a compatible SIP server via the IP network. The built-in switch with downlink function allows direct connection of an additional IP device (e.g. an IP camera).

Besides high volume, the SIP station provides a numerous amount of further features: Pre-recorded audio can be applied in a multi-purpose manner, e.g. as acoustic indication at line fault or as waiting information at call initiation. A configurable background noise suppression provides a crystal clear communication in challenging situations.

Furthermore, the station is perfectly suited for use as door stations at entrances and gateways, due to integrated relay outputs.

The robust construction provides full protection against water, dirt and dust – IP rating IP66.

Features and highlights

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 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

Automated voice messages

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.

Always at your service, thanks to redundancy

- 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

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

Relays 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).

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.

Location identification messages

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

This feature 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

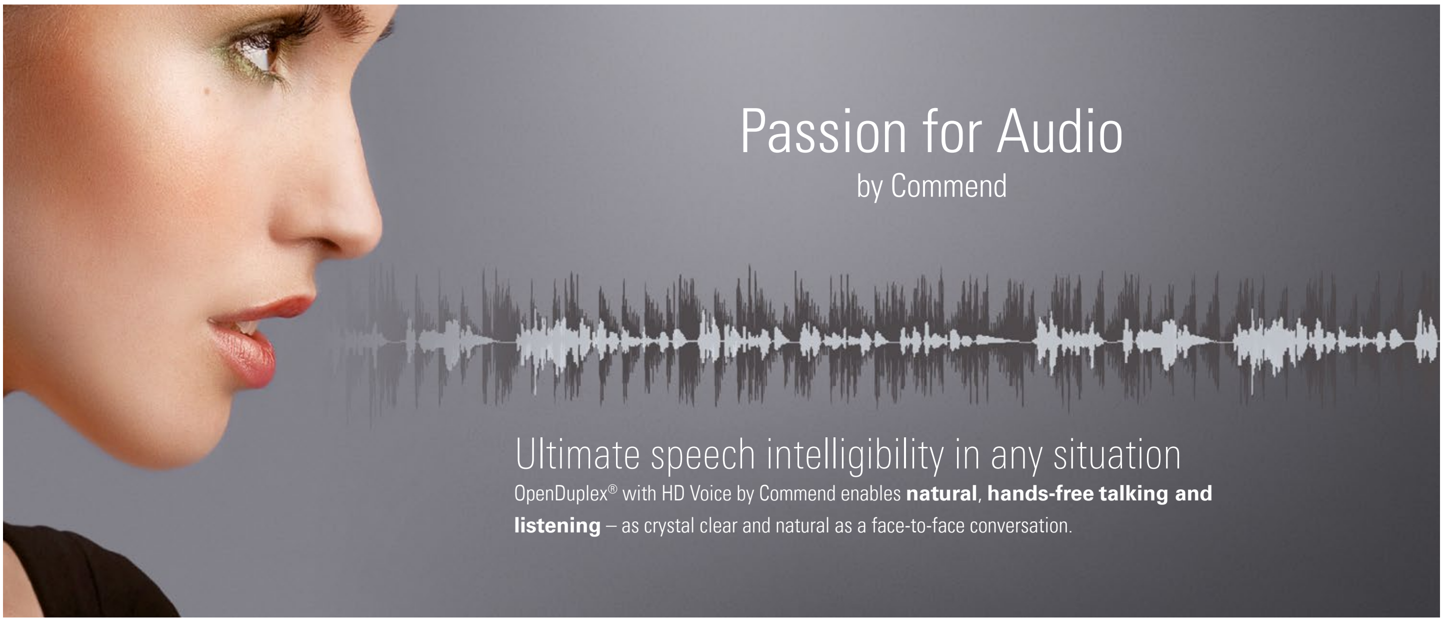
The stations are specifically designed for easy, convenient configuration over the special web interface. A few clicks is all it takes to perform an update and even set up complex action sequences. For large-scale installations, the provisioning function helps to deploy configuration settings automatically 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 adding features such as announcements, audio recording, interfacing with external systems (e.g. visualisation), and many more.

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



The ORIGINAL OpenDuplex® since 2003
Natural communication

7 kHz HD Voice
Crystal clear audio

High volume

Background noise suppression

Loudspeaker/microphone surveillance

Audio // Basics

HD Voice	HD Voice by Commend transfers the audio signal at a bandwidth of **7 kHz**
Sound pressure level	**High volume** up to **99 dB**
Amplifier	High efficient class-D amplifier with 2.5 W
Microphone	Omnidirectional electret condenser microphone for max. 7 m (23 ft) speaking distance
Loudspeaker	2 x 8 Ω loudspeaker with humidity-resistant special membrane type for optimum sound quality

Learn more
audio.commend.com

Audio // Functions

Dynamic **background noise suppression** virtually eliminates all ambient noise

Loudspeaker/microphone surveillance – ensures the availability of the Intercom station while reducing the need for manual verification of its functionality

Peer-to-peer audio – reduces network and server load to ensure efficient use of resources

Conference call function for simultaneous talking with multiple conversation partners

OpenDuplex® for natural, hands-free communication

Switched Duplex for situations with extreme ambient noise (e.g. tunnels)

SIP-WS 800P
Technical Specifications

IP rating acc. EN 60529:	IP66
Front panel:	polycarbonate
Operating temperature range:	–20 °C to +70 °C (–4 °F to +158 °F)
Storage temperature range:	–20 °C to +70 °C (–4 °F to +158 °F)
Relative humidity:	up to 95%, not condensing
Keypad:	alphanumeric full keypad, white backlight activation force: 3 N, 1 x 10^6 cycles
Display:	monochrome LCD display, 128 x 64 pixels, white backlight
Microphone:	electret condenser microphone, polar pattern: omnidirectional, speaking distance: max. 7 m (23 ft)
Loudspeaker:	special membrane type for optimal sound quality, sound pressure level: 85 dB/1 W/1 m (3.28 ft), 2 x 8 Ω
Amplifier:	built-in class-D amplifier with 2.5 W
Sound pressure level:	max. 99 dB
Handset, headset:	EM sensitivity: 14 mV_{eff} EM impedance: 3.3 kΩ, EM supply: 2.5 V EP level: 850 mV_{ef}f at 0 dBm0, EP impedance: 200 Ω
Status indication:	multifunction LED (colours: red, green, blue)
Line output:	for connection of loudspeaker module
Outputs:	2 relay outputs (switch-over contacts) max. 60 VDC, 2 A, 60 W [1)] expected life: min. 5 x 10^4 (2 A), 10^5 (1 A)
Inputs:	3 inputs for floating contacts
Protocols:	IPv6 ready, IPv4, TCP, UDP, HTTP (RFC 2617, RFC 3310), RTP (RFC 3550), RTCP, DHCP, SDP (RFC 2327), SIP (RFC 3261), SNMPv2, STUN, TFTP, URI (RFC 2396), DTMF Decoding (RFC 2976, RFC 2833), SIP User Agent (UDP RFC 3261),SIP Refer Method (RFC 3515)
Transmission bandwidth:	7 kHz
Connection:	pluggable screw terminals expansion jack for e.g. EB2E2AHE IP uplink/downlink: shielded RJ45 modular jacks
Cabling:	min. Cat. 5
Audio features:	OpenDuplex®, Switched Duplex background noise suppression, pre-recorded audio
Power supply:	24 VDC ± 2 V, 500 mA or PoE power consumption: 1.6 W idle approx. 2 W at conversation (depending on volume)
PoE (Power over Ethernet):	following IEEE 802.3af power consumption of the terminal device: class 0 (0.44 W to 12.95 W)
Codecs:	G.722, G.711 a-Law, G.711 µ-Law
ONVIF specification:	ONVIF Profile S for unidirectional audio
System boot time:	within seconds
Ethernet:	2 x 10/100 MBit/s (Full/Half Duplex)
Additional mounting material:	flush mount kit WSFB 50P surface mount kit WSSH 50P
Dimensions (W x H x D):	with flush mount kit: 165 x 280 x 13 mm (6.5 x 11 x 0.51 in) with surface mount kit: 165 x 280 x 51 mm (6.5 x 11 x 2 in
Weight incl. package:	approx. 750 g (1.65 lbs)

[1)] The relay output may only be connected to a SELV circuit! A SELV circuit as per IEC/EN 60950-1 must be separated safely from a dangerous electrical circuit (e.g. 230 V or 110 V mains power), e.g. by means of double insulation. The SELV circuit must not exceed 60 VDC or 42.4 VAC_{peak} (30 VAC_{eff})!

Line length in LAN

The maximum line length of Cat. 5 cabling in a LAN is 100 m (328 ft) – e.g. from switch to the SIP station.

Extent of supply

- SIP station
- Device identification document
- Short reference

System requirements

- Compatible SIP server (see TE | 2) or
- VirtuoSIS (min. PRO 800 5.0, min. base licence PRO 3) or
- GE 800 with G8-VOIPSERV or
- Serverless operation

Compatibility SIP PBX

Basically, the SIP stations can be used with any SIP server.
The following server types have been tested explicitly by Commend International GmbH and therefore a proper functionality can be confirmed:

Hersteller [3]	Typ	Version
Cisco	Cisco Call Manager Cisco Unified Communication Manager	Version 5, 6, 7, 8
Digium	Asterisk	Version 1.2, 1.4, 1.6
Avaya (former: Nortel)	CS1000	Version 6
Avaya	Avaya Aura™ (Avaya Communication Manager, Avaya Session Manager)	Version 6.1
Innovaphone	Virtual Appliance IPVA	Version 9 final
Alcatel	OmniPCX Enterprise (OXE)	Release 9
Siemens	Hipath 4000 Hipath 3000 + HG 1500	Version 5
3CX	3CX for Windows	3CX Phone System Version 9, 10, 11
Starface	Starface free	Version 4.x, 5.x
Aastra (former: Ericsson)	MX-ONE	Version4.1 SP 1
Kamailio	Kamailio (OpenSER)	Version 3.3.0
FreeSWITCH	FreeSWITCH	Version 1.1 Beta1
ELMEG	elmeg ICT880	Version 7.67D
2N®	2N® Netstar IP	Version 3.10.96
AVM	Fritz!Box Fon 7170 Fritz!Box Fon 7270	Version 29.04.87 Version 54.05.05
Sipgate	sipgate.at, sipgate.de	getestet Dez. 2010
Vodafone Arcor	vodafone.de	getestet Jan. 2011
blueSIP	blueSIP.net	getestet Mai 2011
Mitel	3300ICP	12.0.0.49

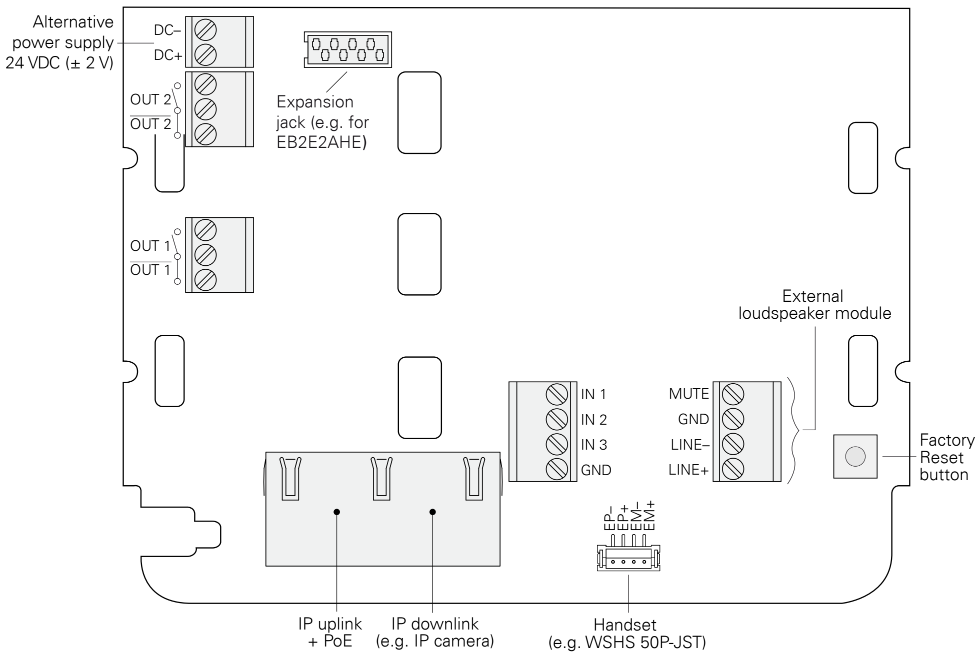
[3] The listed products and company names are brand names or registered trademarks of their respective owners.

SIP-WS 800P
Installation Instructions

Precautions

- When opening the stations, ESD precautions must be observed.
- The stations may only be opened by authorised service engineers.

Connection



Attention:

- It is mandatory to ensure a correct power supply of the SIP station: min. 22 VDC, max. 26 VDC.
- PoE has to be connected to the RJ45 jack “IP uplink” (see above).
- Further IP devices can be connected to the RJ45 jack “IP downlink” (see above), like an IP camera. But it is also possible to link up to 20 SIP stations in series via this Ethernet interface.
- Use of PoE: If multiple SIP stations are connected in series, then only the first device can be supplied with Power over Ethernet. All other devices (connected in series) must be supplied separately by an external power supply unit.

Configuration via web interface

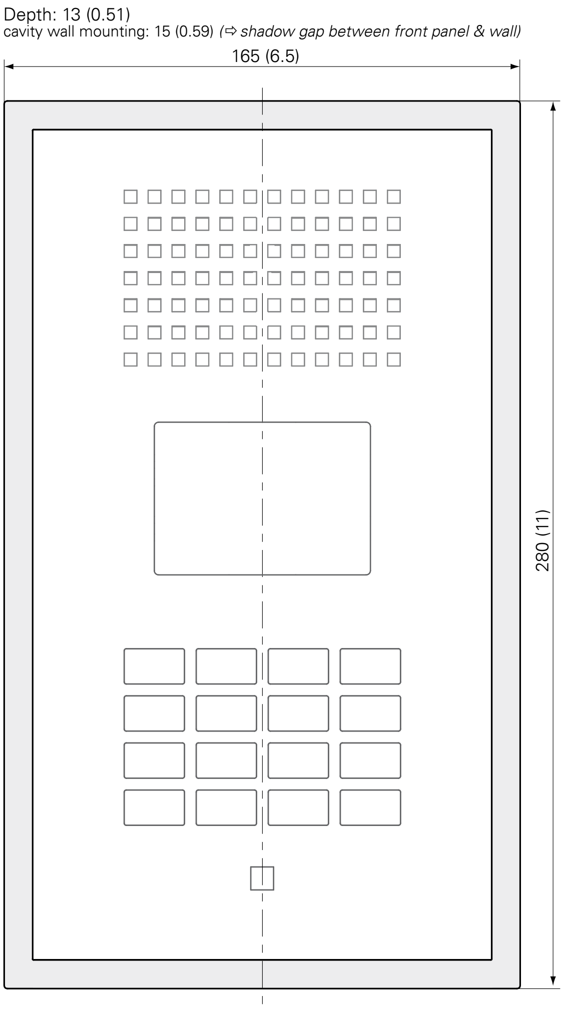
Via the integrated web interface, it is possible to configure optional direct dialling numbers for the buttons of the SIP station.

Mounting instructions

- Do not expose the station to extreme temperature (see "Technical Specifications" on TE | 1).
- For flush mounting, a flush mount kit WSFB 50P (available separately) is required.
- For surface mounting, a surface mount kit WSSH 50P (available separately) is required.

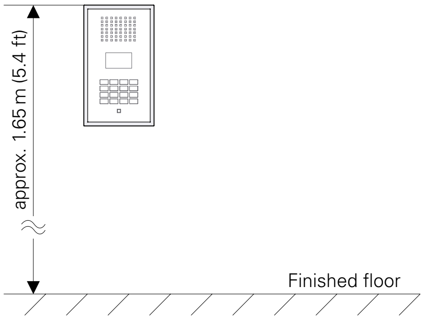
Dimensions front panel

Measuring units in mm (inch), not to scale!



Recommended mounting height

The upper edge of the station approx. 1.65 m (5.4 ft) from the finished floor. Please adapt the mounting height to the individual needs.



Quality tested. Reliable. Smart.

COMMEND products are developed and manufactured by Commend International in Salzburg, Austria.

The development and manufacturing processes are certified in accordance with **EN ISO 9001:2015**.

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A strong worldwide network

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

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