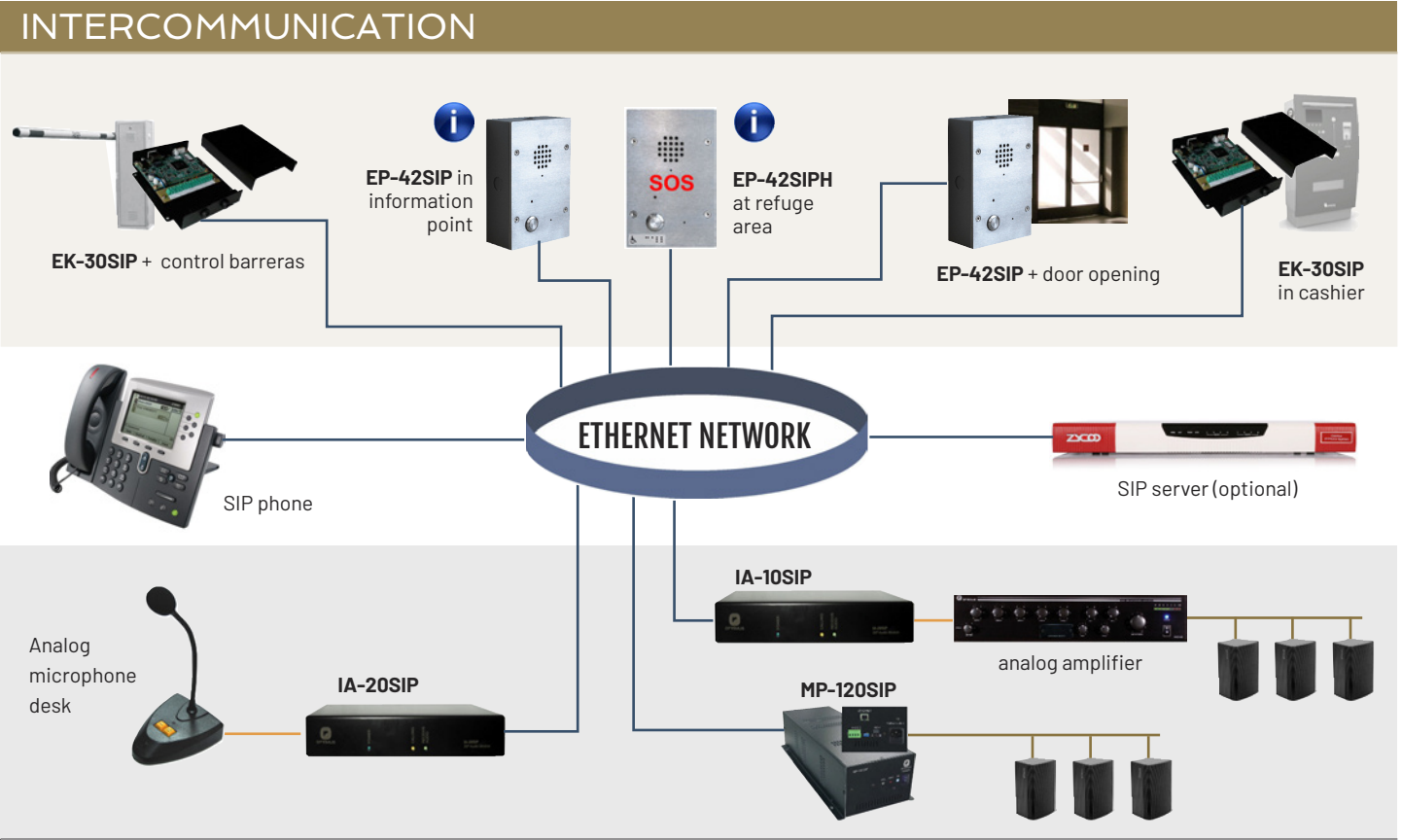


IP - SIP SYSTEMS

INTERCOMMUNICATION & PUBLIC ADDRESS



PUBLIC ADDRESS

SIP PROTOCOL

Control and signalling protocol mainly used in **IP telephony systems**.

It allows to initiate, modify and terminate communication sessions with one or more participants (intercom, phones and interfaces...).

It enables **compatibility** between devices and **integration** in global control systems.

OPTIMUS Systems with SIP protocol eases configuration for control and communication systems at a great distance.

Suitable in...

- access areas
- doors
- stairs
- barriers
- elevators
- ATM
- tolls
- suppliers
- information
- help point



INTERCOMMUNICATION & PUBLIC ADDRESS

INTERCOMMUNICATION

EP-42SIP | EP-42SIPH

Vandal-resistant and weatherproof intercom station with SIP protocol. Bi-directional full duplex audio, echo cancellation, noise reduction, web server, metal call button, led for active call, remote control for door unlocking, signalling output to activate external Systems.



The **EP-42SIPH** is a special model for emergency call points, such as those installed in the refuge areas, with large SOS letters in red, pre-recorded message for call in progress, accessibility pictogram and CALL word in braille.

EK-30SIP

IP/SIP Intercom electronic board in metallic box, with speaker, microphone and push-button, to integrate into ATMs, elevators, barriers...



Bi-directional full duplex audio, echo cancellation, noise reduction, web server, remote control for door unlocking, signalling output to activate external Systems.

MECHANICS

EP-42SIP | EP-42SIPH

Sturdy 3mm metal front panel

Mylar speaker (**weatherproof**) and protected by an inner metal grille

Metal **vandal-proof** call button

Security **Torx** Screw

IP65 protection against dust and water

AUDIO

Most common **codecs**: G711, G722, G726, DVI4, linear PCM, Speex... included

Noise reduction

Echo cancellation

Ringtone in process customizable by WAV file

Door open acoustic alarm

Microphone and speaker **volume** setting

CONTROL

Temporary **door opening** from zone control using DTMF code

DTMF codes to **leave the door open or closed** permanently.

Additional contact, active in communication or call, to control external systems such as CCTV

CENTRAL CONTROL

One of the greatest advantages of using SIP protocol is the possibility of using any SIP phone as call monitoring station. Please, contact us for further information about the most suitable system to your requirements.

OPTIMUS has also specific control stations to take full advantage of the system's capacity.



SOFTWARE

Configuration can be done from any **browser** by using the web server built into each unit. With **DHCP** (Dynamic Host Configuration Protocol) server to obtain automatically IP address and Protocol **SSDP** (Simple Service Discovery Protocol) to appear in the view of network Windows 7 or higher. Consecutive screens provide a quick and easy configuration of IP-SIP systems.

The image displays four screenshots of the web configuration interface for IP-SIP systems:

- NETWORK screen:** Shows WAN settings including General, Static IP Address (193.254.71.84), and STUN settings.
- SYSTEM screen:** Shows System configuration options like Backup/Download, Restore/Upload, and Firmware version (1.4.4.20140131).
- AUDIO screen:** Shows Intercom Audio settings (Speaker volume, Microphone volume) and Codec Selection (G711 ulaw, G722 HD, etc.).
- ACCOUNTS screen:** Shows Account configuration options (General, Topology, QoS, Advanced) for adding accounts to connect to a PBX.

SERVERS

U-20 | U-50 | U-100

Servers enhance the performance of intercom and PA interfaces and allow to configure systems with additional functions to those based on point to point p2p.

Among other features, servers allow audio recording, output to analogue DTMF telephones, use of generic SIP smartphone applications, running behind a NAT (in different networks without VPN), sequential call, multiple call, call paging, use of generic SIP smartphone applications (simultaneous audio) simplex and duplex...



- 30 / 100 / 500 extensions
- 15 / 30 / 100 concurrent calls
- Codecs G711, G722 HD, G726...
- Hardware echo cancellation
- Modular design for flexible solutions
- Web server
- No need for software licenses



U-20 | 30 extensions | 15 concurrent calls



U-50 | 100 extensions | 30 concurrent calls



U-100 | 500 extensions | 100 concurrent calls

IP - SIP SYSTEMS

INTERCOMMUNICATION & PUBLIC ADDRESS

PUBLIC ADDRESS

IA-10SIP | IA-20SIP

IP/SIP audio interfaces to analog audio. They allow to transmit audio from any SIP device to an analog public address system: amplifier, preamplifier, power unit...

The model **IA-20SIP** is bidirectional, it also allows to connect an analog audio source (microphone, musical source ...) and transmit the signal to SIP devices.



- Parameter configuration via WEB server.
- Local power supply 5 Vdc (I905SIP) oPoE.
- Direct audio output for loudspeaker (8 Ohm, 2 W).
- AUX audio outputs (0 dB) and MIC (-60 dB).
- Relay contact output (activation by DMTF).
- Open collector priority output (activated in connection).
- Indicators: power supply, call and audio level.

IA-20SIP only

- Audio input with selectable level, AUX/MIC.
- Input contact for call (intercommunication) or priority (PA).
- Operating mode: broadcaster, receiver or bidirectional.

MP-120SIP

Digital power unit class D with speaker output for 100V line, volume control, protection against tension peaks, short circuits, overheating and overloading protection, led indicators for power and protection.

IP/SIP input and additional 0 dB output to connect with other power units.



Power supply	230 Vac, 50 Hz
Nominal power	120 W
Loudspeaker output	100-V
Audio output	775 mV / 0 dB
Frequency response	80 ~ 18,000 Hz
Signal-to-noise ratio	> 85 dB
Harmonic distortion THD	< 0.1%
Ethernet	10 / 100 Mbps
Consumption	150 W
Dimensions (mm)	132 x 92 x 385
Weight	2.4 Kg



SECURITY



INFORMATION



MUSIC



COMMUNICATION



PAGING



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