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Filiali a Roma e Venezia-Mestre

TASVoIP-95x Digital Emergency / Info Call Telephone Point

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SISTEMA QUALITÀ CERTIFICATO ISO 9001
ISO 9001 CERTIFIED SYSTEM QUALITY

TASVoIP-95x / SOS-Emergency

- ***Digital VoIP telephone (standard SIP) for flush mount installation built on weatherproof IP65 vandal-proof aluminium plate***
- ***Hands-free conversation***
- ***Large mushroom head direct call keys***
- ***Operating Temperature: -25°C ÷ +70°C***
- ***Available with 1 or 2 direct call keys***
- ***RJ45 Ethernet interface with PoE and/or local power supply (12 ÷ 56 VDC)***

The station has been designed for meeting the customer's requirement for handling the Emergency and Information calls in transport and heavy industrial environments applications.

The TASVoIP digital telephone series are built on aluminium plate designed in order to guarantee the IP65 (IEC60529) protection degree. The telephone is delivered complete with the relevant GRP weatherproof IP65 rear box including the flush mounting gaskets.

The digital TASVoIP-95x series is available also with the special **IL** *(Inductor Loop)*.
In this version, the station is assembled with the electronic board and the inductive antenna designed for radiating the audio signal as inductive waves.
Thanks to this, the **hearing-impaired person** equipped with the proper acoustic hearing aid can establish the conversation with the operator.
The digital TASVoIP-95x, equipped with the Inductor Loop equipment, requires the additional power supply from 12 up to 56 VDC / 20 W.

The 95x series is available also with analog technology (the series is named TASVOX) in order to allow the direct connection of the station to the PSTN and/or the PABX exchange.
Also the TASVOX analog series are designed for allowing the remote diagnostic facility.

Accessories

- Piggy-back board equipped with 2 relays for remote-control activities (for example, open-the-door, switch ON/OFF the light, switch ON/OFF the CCTV camera, and so on); each relay is equipped with one change-over contact. In addition, the piggy-back board is equipped with n. 3 opto-isolated inputs
- "Inductor Loop" piggy-back board .

[The use of accessories may include the need for additional power supply and/or checks for the contemporaneousness of the required functions]

Technical Characteristics	
Front plate	4 mm. thick 50-60 copper-free aluminium (IP65 weatherproof protection degree)
Rear protection through the weatherproof IP65 box made in orange “flame retardant” GRP material	- Flame Class rating UL94-V2 - Flame class rating UL94-V0 - Rating UL94-5VA - Glow Wire Flammability Index 960°C IEC60695-2-12 - Oxygen index (LOI) ISO4589: 32%
Fixing	Flush mounting
Input cables	n. 2 weatherproof PA6 polyam.M16x1,5 cable glands (cable: $Ø_{min}$ 5 mm. – $Ø_{max}$ 10 mm.)
Key dimension	Ø 25 mm.
LED for visualising the line status	YES
Hands-free conversation	YES
Full-Duplex conversation	YES
Trebles and volumes	Programmable through Web-Server pages
Ringing sound power	94 dB @ 1m.
Hands-free microphone	Weatherproof electret
Loudspeaker output volume	Adjustable through Web-Server page: max. 250 mW -
Analog audio output	0 dB level – to be connected to the input of the external amplified loudspeaker
Reference Norms	EN-55022 (2006) + A1 (2007) EN-55024 (1998) + A1 (2001) + A2 (2003)
Operating Temperature	from –25°C up to +70°C
Storage Temperature	from –40°C up to +80°C
Relative humidity	95% non condensing
Dimensions	H 330 x L 195 x P(flush) 70 mm. – (packed: 350 x 240 x 120 mm.)
Weight	kg. 1,350 – (packed: kg. 2,250)
Consumption (max.)	≤ 4 W

VoIP characteristics with SIP protocol	
LAN/WAN connection	10 BaseT Ethernet RJ45 with UTP cable – IP static address or DHCP
PoE (1)	YES – through RJ45, according to the 802.af norms
External power supply (1)	from 12 up to 57 VDC
Communication protocol	SIP
Audio codecs	G.711 A-low – G.711 µ-low – G.722 – G.729
Remote configuration with protection through password	➢ Web server ➢ Download of the configuration file ➢ SNTP with local time and day light saving time handling ➢ Updating through TFTP, FTP, HTTP ➢ Configuration uploading file ➢ VLAN
Option: I/O Piggy-back module (3 IN and 2 output relays)	- n. 3 opto-isolated inputs (it is necessary to provide the external power supply) - n. 2 relays, each equipped with one change-over contact with V_{max} 250VAC – A_{max} 2A – P_{max} 62 WAC / 30 WDC
Remote real-time diagnostic functions	• Through ASTRO System Server configured con ASTRO/Manager (including VNC) or • Through Web Server or • Through POST

(1) *NOTE*: The digital FITRE stations are designed for accepting simultaneously the power supply coming from PoE and from the local power supply in order to guarantee the operation even in case of fault on one of the two power sources

Functional Characteristics

- The station is available in following versions:
  - TASVoIP-951 equipped with 1 direct call key (also in special Inductor Loop version)
  - TASVoIP-952 equipped with 2 direct call keys (also in special Inductor Loop version)

- Each key is independent and it can be configurable in order to activate automatically the telephone number dialling

- *Option*: Multi-functions keys, address book, call group.
  Each key can be configured for activating:
  - a specific function; for example: direct call, call on condition, dial-tone (DTMF) generation, and so on
  - a number of the address book
  - a group call; for example, for calling simultaneously several operators

- Hands-free – real full-duplex conversation

- Automatic self-answering after a programmable time

- Automatic hung-up when the operator hung-up own handset

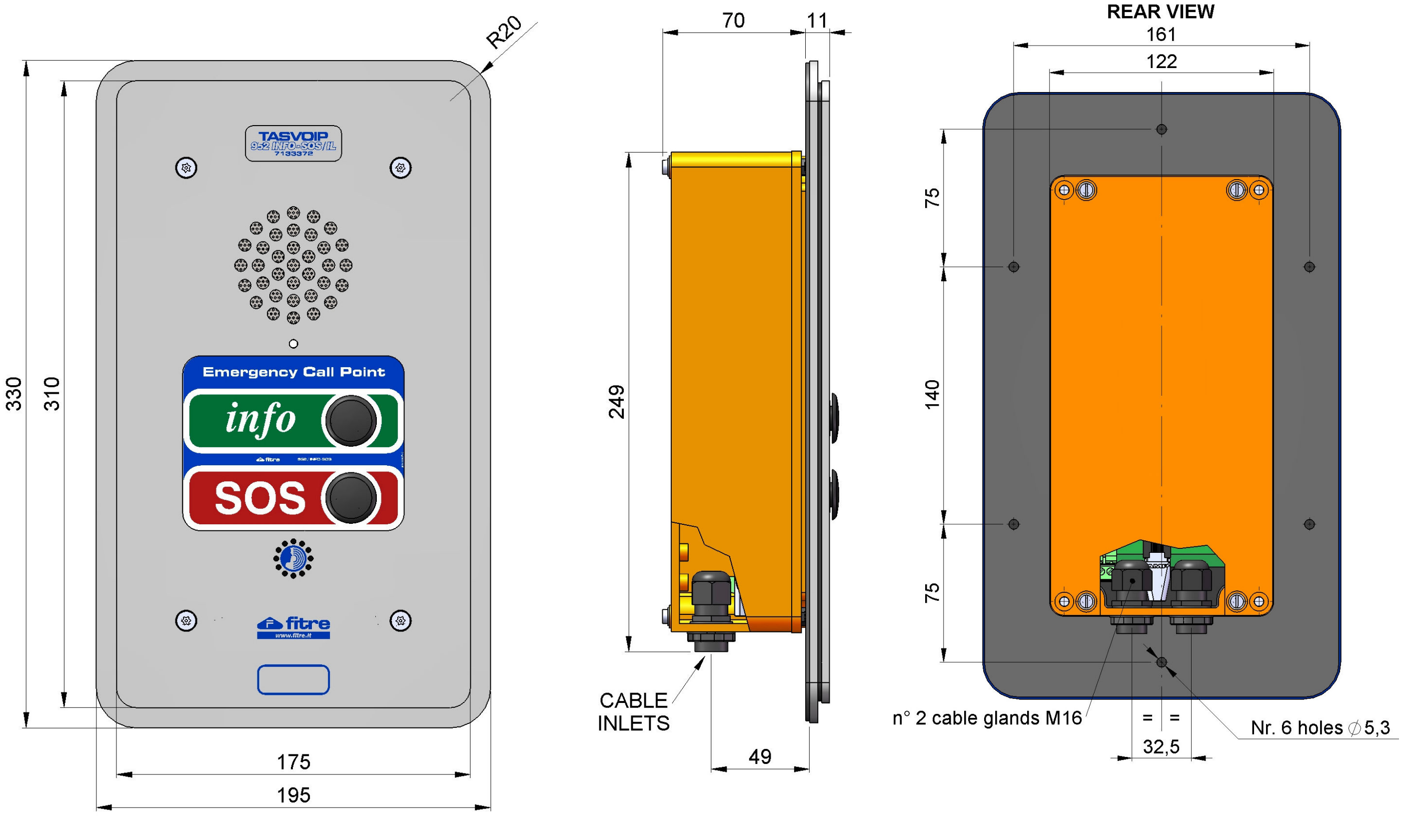
- Signalling LED for visualising the status (call in progress)

- “SOS” - “INFO” identification tag

- Remote diagnostic facility through password. The diagnostic function includes the checking of the:
  - Line status (call receiving)
  - Configuration data, stored inside the telephone itself

- Available for “transport” applications (railway, metro, tramway, and so on) where the “silent monitoring” function is required

Dimensions:



Configuration through browser web

The TASVoIP 952 INFO/ SOS includes a web server allowing an easy and fast configuration of the telephone set.
When the telephone is connected to the LAN that gives the , knowing the IP address, it is possible to access immediately to the relevant web server using a simple browser just after the switch-on.
The TASVoIP web server offers the possibility for configuring several data and facilities/functions using different pages:

"Main Page"	This is the main page which can be accessed after the login and it includes three different fields: 1. a list of all the possible lines, 2. an interface to call another device, 3. a report of the current calls.
"Call Info"	This page shows the history of the incoming/outgoing calls of the telephone. The calls are listed according to the date and time they were made
"Line login"	Configure the SIP line with the relevant extension number (account) on the SIP registar
"Line NAT"	Specific a NAT server; typically, to provide a translated connection to Internet
"Line RTP"	Select the audio formats acceptable by the telephone set; in addition, it is possible to configure the priority level list for using the audio formats.
"Address Book"	List of subscriber and group numbers to be saved in the address book in order to speed-up the call procedure
"Multifunction button"	Configure the multifunction buttons, in order to link a specific action to each key, activated automatically by the telephone when the user presses the function key. The number of available function keys depends on the model of the telephone.
"Advanced"	Set-up of some advanced options related to: • Network configurations • Busy tone timeout • Handling of the IP and MAC-address information reading • Handling of the web server access
"Preferences"	- Audio setting (ON/OFF) - Ringer (select the ringer tone choosing it on the available list) - Intercom & paging (enable/disable the receiving of intercom/paging voce messages/tones) - Spy setting (enable/disable the spy calls reception) - DTMF modality (selection of the modality for transmitting the dual-tones) - Auto answer setting (Allows enabling the auto-reply mode and specify the time when the telephone will automatically answer to the incoming call) - Reject timeout on phone call (Time interval during which the telephone rings if an incoming call is present. After this time interval, the call is automatically rejected - Real-time transport protocol (Rtp packets waiting time before starting to consider the connection invalid. In case it is disabled, the connection is always considered valid - Tones (Allows to configure the tone scheme to be used. It is possible to add personalized tones, too. Once saved, these tones will be displayed in the list.) - Audio levels – Audio levels adjusting: • Output Volume on head-set (when the telephone is equipped with the relative jack) • Output Volume when the telephone line is engaged • Output Volume of the external auxiliary device (typically, the amplified loudspeaker) • Output Volume of the hands-free circuit • Output Volume of the calling signal (equivalent to the ringing tone)
"Custom tone scheme"	Define a personalized scheme of tones to be associated with the different telephone states
"Echo canceller"	Allows to change the original configurations from the echo eraser
"Software up-date"	Allows the firmware updating. The file can be uploaded using the ftp server or the http protocol, including the checking of the uploaded file.
"External device"	Configure the auxiliary I/O piggy-back board through a useful wizard help. The optional I/O board is available only on some Fitre telephone VoIP models
"Speech test configuration"	Allows to access the configurations to analyse the available audio devices (including the microphone and loudspeaker units)
"System information"	Get the information about the model, the MAC address, the IP address, the firmware version and the status of the SIP lines configured in the digital telephone set
"Diagnostic"	Get the information about the diagnostic events of the telephone as well about the status of the registration of the configured SIP lines. The data are required in cyclic way and the page is automatically refreshed.
"Packed capture"	Capture the data on the LAN (from / to the telephone set in a programmable time window).
"Setting"	See the current telephone set configuration