FE-CUBE-PAS1 – PRODUCT DATASHEET

PAGING AUDIO ADAPTER AND SCHEDULER FOR TRANSFORMING SIP-CALL TO ANALOG AUDIO SIGNALS

FE-CUBE-PAS1 is SIP to analog audio adapter/gateway able to connect to any SIP-server as standard SIP-client and get a phone call as standard SIP-extension and translate it to analog audio and deliver it to any audio amplifier or active speaker through AUX interface.

Tones and messages can be streamed live or uploaded to device in order to play them later or periodically from build-in timer/scheduler.

Auto-provisioning option through IP-INTEGRA SIP-server saves time, especially in the case of large-scale installations.

The module can easily integrate in 3rd part SIP-systems through WEB interface. Small size allows to place it physically behind or inside active speakers or behind any audio amplifier.

At a Glance

- For integrating analog audio paging systems (active speakers and large amplifiers) with SIP servers
- Paging to one speaker or active speaker groups
- Easy upgrade analog public audio (PA) system to SIP/IP PA
- Standard line-out for any amplifier
- Power voltage: 5V DC, Power Wattage less than 0.5W average
- Audio out: AUX
- Protocols: G.722, PCM/G711 ULaw, GSM

TECHNICAL SPECIFICATION

HARDWARE	
CPU	H2 Quad-core Cortex-A7 H.265/HEVC 1080p
Memory (SDRAM)	256MB DDR3 SDRAM
Onboard storage	TF card (Max 32GB) / SPI Flash
Onboard network	10/100M Ethernet RJ45
Audio input	MIC integrated

Power source	USB OTG can supply power
USB 2.0 Ports	One USB 2.0 HOST, one USB 2.0 OTG
Buttons	Power Button
LED	Power LED & status LED
CONSTRUCTION	
Product size	48 x 46mm
Weight	26g
NETWORKING AND PROTOCOLS	
Protocols	IPv4 (with DiffServ), SIP, TCP, UDP, HTTP, RTP, DHCP, SNMP, NTP
LAN protocols	Network access control (IEE 802.1x)
Management and operation	HTTP (Web configuration) DHCP and static IP, remote software upgrade, centralized monitoring
SIP support	RFC3261 (SIP base standard), RFC3515 (SIP refer), RFC2976 (SIP info), SIP using TLS, RFC5630 SIPS URI scheme
DTMF support	RFC2833, 2976 (SIP info)
SOFTWARE	
Web-interface	Dashboard with HTTPS based configuration, alarm settings
Sound adjustment	Echo-cancel, sound mute, silence suppression, paging start/stop tone, microphone boost/mute
Scheduler	Bulk adding, automatic fill-out, ring-tone upload
Auto – provisioning	Via HTTPS, automatic synchronizing of SIP-server scheduler and sound files
Logging	Call, event and scheduler logs

FREUND ELEKTRONIK A/S, IN COOPERATION WITH ITS SISTER COMPANY FREUND ELEKTRONIKA SARAJEVO, IS DEVELOPING IP-BASED INTERCOM, AUDIO, ACCESS CONTROL ADN SMART HOME SOLUTIONS.

WE HAVE MORE THAN 30 YEARS OF EXPERIENCE IN THIS TECHNOLOGY FIELD AS DEVELOPER, MANUFACTURER AND RESALER COMPANY.

IN THE INDUSTRY, WE NEGOTIATE THE MOST INNOVATIVE SOLUTIONS WITHIN BUILDING COMMUNICATION. WE HAVE DAILY FOCUS ON DEVELOPMENT AND USER-FRIENDLINESS AND OUR PRODUCTS HAVE TECHNICAL HIGH QUALITY AND PLEASENT DESIGN.

AS A DEVELOPER AND MANUFACTURER OF OUR OWN IP-INTEGRA SYSTEMS, WE ARE ON TOP OF THE MOST ADVANCED SYSTEMS FOR DOOR TELEPHONY, PUBLIC AUDIO AND ACCESS CONTROLL SOLUTIONS.

OUR DEVELOPMENT DEPARTMENT HAVE DEVOLPED, TOGETHER WITH OUR PARTNERS, ELEGANT AND ROBUST DOORPHONES, SIP-CENTRALS, TERMINALS, IP-SPEAKERS, ACC-CONTROLLERS AND APPLICATIONS WITH INTELLIGENT FEATURES AND THE MOST ADVANCED TECHNOLOGY, BUT AT THE SAME TIME EASY FOR OUR CUSTOMERS TO USE.

FREUND GROUP IS ONE OF THE TOP PROVIDERS ON THE MARKET FOR IP-BASED BUILDING CONTROL AND COMMUNICATION AND RELATED SYSTEMS.