



Public Address - Voice Alarm

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Commercial Audio

Intelligent Acoustic Solutions

Loudspeakers



CATALOGUE 2011

Commercial Audio

DELIVERING YOUR MESSAGE



Public Address - Voice Alarm
IPVA - Intercom
Commercial Audio
Intelligent Acoustic Solutions
Loudspeakers



Dear Friends,

Following the new PAVA catalogue, once again we proudly present to you our new section, covering;

Commercial Audio

This new impressive portfolio has been compiled in a whole new format showing you our vision on product designs and product innovation. As you go through the pages, you will discover that our portfolio has undergone many changes compared to our previous edition.

Audio networking has become a very common item nowadays and most of the powerful audio routers are equipped with it. Unlike most manufactures, who chosen the simple solution, ATEIS chose to develop its own low-latency audio and data network devices that are capable of handling up to 48 broadband audio streams with a latency of less than 1 ms. This makes the ATEIS-NET systems perfectly adapted for live-performance applications. Besides the networking, all our routers are equipped with very powerful pre and post processing elements, paging facilities, teleconferencing and VPN connectivity and the list continuous ...

Please take your time to study this collection of valuable information and ensure yourself that ATEIS delivers the right products and the best flexible solutions you have been looking for.

Take the opportunity to share your thoughts with us and we will provide you the solution!

We are looking forward to your business!

Team ATEIS is

DELIVERING YOUR MESSAGE!



INDEX

LAP DSP audio matrix System

Designed for large sized pro-sound applications, the LAPs are the first products to combine PRO-sound requirements within a multi channel secured and dedicated low-latency audio- and data network.

LAPG2 / Networked Linked Audio Processor

NET-Lx / ATEIS NET secured Audio Network

UAP DSP audio matrix System

Specially designed for small to medium Commercial applications, the UAP is the new DSP audio matrix for medium paging and multizone audio routing applications.

UAPG2 / Expandable Universal Audio Processor

ECS Echo Cancellation System

The ATEIS ECS-Teleconference-unit, is a digital signal processor featuring DYNAMIC Automatic Echo Cancellation based on our own developed wideband acoustic and line-echo cancellation algorithm. Designed specifically to provide clear audio in teleconferencing applications.

ECS / Networked Echo Cancellation System

Consoles and Accessories

RAC5 / RAC8 / URC / URC200 / NSM / Controller Devices

PPMIT5 / PPMWJB-V3xx / PM1 / Paging Consoles

3

4

5

6-7

8-9

10

11

T

Z

E

T

Z

O

C



LAPG2

Networked Linked Audio Processor



Easy to use PC software for system design and control (GUI)



LAPG2 Features

- Free DSP architecture.
- CAT5 and fiber optic redundant audio networking capabilities.
- Internal processing of audio signals can be fully programmed to suit the client's application.
- Excellent sound quality (24 bits, 48 KHz and 96 KHz sampling).
- Impressive array of signal processing tools.
- Easy to use PC software for system design and control (GUI).
- Advanced Preset manager.
- Powerful microphone paging and remote control functions.
- Highly flexible input and output configurations.
- Two 600 Mflops DSPs.
- Up to 32 LAPs on the network.
- Latency < 1 msec.
- Up to 32 microphones per LAP.
- Up to 32 remote controllers per LAP.
- 100 V and Low impedance surveillance.
- EN 60849 and BS 5839 compliance.

LAPG2

LAPG2 - 4In12Out

4 Inputs - 12 outputs Audio processor

LAPG2 - 8In8Out

8 Inputs - 8 outputs Audio processor

LAPG2 - 12In4Out

12 Inputs - 4 outputs Audio processor

LAPG2 - 16In

16 Input Audio processor

LAPG2 - 16Out

16 outputs Audio processor

Designed for PRO Audio and Commercial applications, the LAPG2 Networked linked Audio processor are the first products to combine secured networking and PRO-sound requirements.

Sonic excellence

The advanced 24 bits A/D and D/A converters, together with the 96 kHz-capable audio processing and the 400 mHz SIMD SHARC core, capable of 2.4 GFLOPS peak performance, guarantee an excellent sound quality and low latency.

Impressive array of signal processing tools

The LAPG2 are comprehensive systems which integrate pre-amplifier, compressor-limiter, equalizer, as well as matrixing and delay functions into one unit. Useful features like Automatic Gain Control, Feedback killers, Automatic Microphone mixers and Crossovers are also part of the LAPG2 DSP components library.

This new generation provide a message storage component which able to store several audio message in the LAPG2.

The following events: Play a message, change master preset, sub preset, element adjustment or set the TTL out can be controlled by third party protocol, by an analog input or by the scheduler. The scheduler can lead all the events described above. Internal processing of audio signals can be fully programmed to suit the client's application.

Installers can select the audio processing feature(s) which they wish to apply to the various inputs and outputs from a library on their PC, using software provided with the LAPG2. Once the configuration process is completed, it can be loaded into the LAPG2. All configurations can be backed-up onto PC and loaded into the LAPG2 as and when required.

Advanced Preset manager

The LAPG2 includes two types of presets :

- More than 20 Parameter presets : they enable values of multiple parameters of the same design, such as levels, gains, EQ, etc. to be restored either from the PC software, the remote controllers or the control inputs.
- More than 10 Design presets : they enable completely different designs to be restored. An application example for this feature are hotel meeting rooms with removable walls.

Furthermore, LAPG2 now provides a TCP/IP port with RJ45 connector. PC-based custom control panels can now operate the LAPG2 from remote locations through a TCP/IP network.

LAP System

NET-Lx

ATEIS NET Secured Audio Network



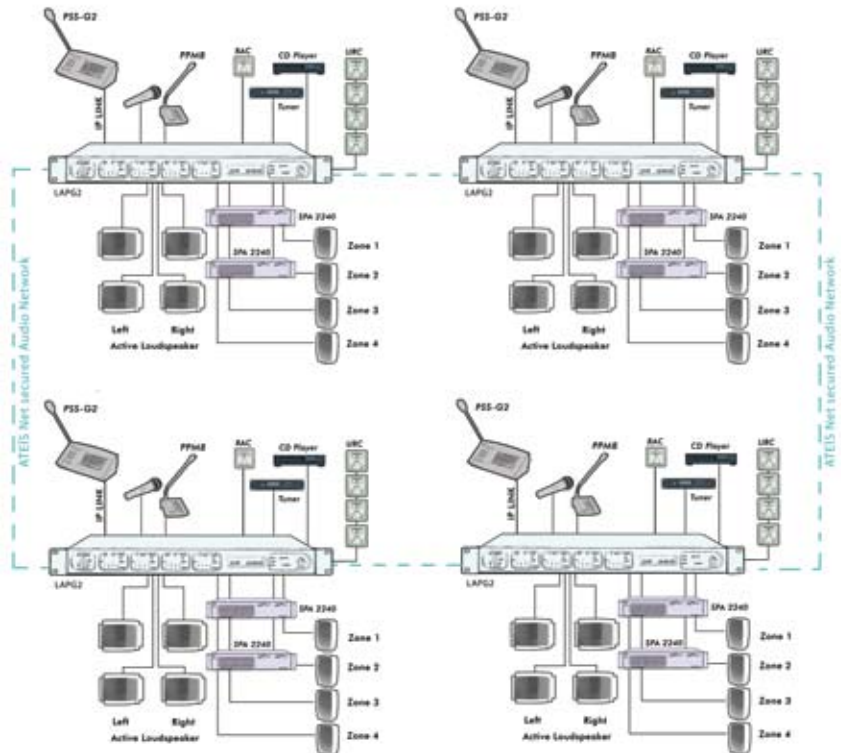
The front panel display and rotary knob allow following setups to be performed without a PC : Time and date settings, preset change, log file access. Of course, those front panel settings can be password protected.

AUDIO CARD INDICATIONS

- Clip: yellow
- Signal: green (with sensitivity selection)
- Phantom: green
- Input: green / Output: green
- AES/EBU: green

GLOBAL INDICATIONS

- Power OK: green
- Fault: yellow
- EVAC: red
- Stand by: green
- Data: green
- Ethernet: green
- ATEIS-NET: green



16 Analogue control inputs

The LAPG2 has 16 control inputs which can be configured as analog control input (0 to 5 VDC or as logic input (TTL)) Each control can be associated to any of the variable audio processing functions of the LAPG2 (input level, output level, equalization, routing). Several parameters (Min + Max values, positive or negative variation, linear, log, anti-log) can be programmed for each of those controls.

16 Logic inputs, 8 Logic outputs (GPIOs)

In addition of the logic inputs, each LAPG2 is also equipped with 8 logic outputs (TTL). Each of those hardware input/output's can be associated to virtually any software button the system designer requires to. The logic inputs could be use to mute or activate an audio signal while the logic outputs enable the LAPG2 to control external equipment. The logic inputs can be used in normal or binary mode.

RS-232 serial interfacing for third party control

The LAPG2 can be controlled from third party equipment like Vity, AMX or Crestron via its RS232 serial port.

CAT5 and fiber optic redundant audio networking capabilities.

ATEIS has developed its own audio networking system "ATEIS-NET Secured Audio Network". This Ethernet based network is able to simultaneously transport 48 audio channels (32 bits, 48 KHz sampling rate) with a latency < 1msec together with the necessary control data.

For decentralised or big applications, an optional networking card can be inserted inside the LAPG2s.

Thanks to its loop architecture, the ATEIS-NET Secured Audio Network audio network is fully redundant. If a problem (Line open or shorted) occurs on a loop segment, it will be automatically isolated without affecting the system functionalities.

Up to 32 LAPG2s can be connected together on the same network. As the Network addresses are auto-negotiated, the network set up is very easy. Once programmed the system will be able to work independently (off-line) without the PC.

NET-L1 / With two RJ45 connectors for CAT5 cabling, max 100 m/300 ft between two LAPs.



NET-L2 / With one ST-Fiber (port A) + 1x RJ45 connectors (port B).



NET-L3 / With two ST-Fiber connectors, Multimode, max 2000 m /6000 ft between two LAPs.



NET-L4 / With one RJ45 connector (Port A) and one ST-Fiber connector (Port B).



UAPG2

Expandable Universal Audio Processor



Easy to use PC software for system design and control (GUI)



UAPG2 Features

- Easy design friendly user control windows for client operation.
- Excellent sound quality (24 bit A/D and D/A converter, 48/96 kHz sampling, SHARC 32 bit-266 Mhz DSP).
- Impressive array of signal processing tools.
- Audio I/O: 4x12, 8x8, 12x4, 16i or 16o, line/Mic and 48 V Phantom power.
- Free drag and drop architecture digital audio processing modules.
- Smart Auto Gain Control (AGC) through background noise level sensing.
- Easy to use PC software for system design and control (GUI).
- Advanced Preset manager.
- Hundreds of presets and an advanced event scheduler.
- Message player (up to 20 minutes in 16 bits and 16 kHz bandwidth).
- Powerful microphone paging and remote control functions.
- Highly flexible input and output configurations.
- 8 programmable remote controls on front panel, and 8 logic output controls.
- 6 programmable remote controls up to a distance of 5 kms via CAT-5.
- Easy setup 3rd party control via RS232 or Ethernet.
- Ethernet and TCP/IP connection for easy system setup and remote control.
- RS485 connection for ATEIS remote devices:
 - URC (Universal Remote Control)
 - PPM (Programmable Paging Microphone)
- Ability to digitally link up to 12 units and share 16 audio channels in 48 kHz or 5 audio channels in 96 kHz.

UAPG2-4In12Out

4 inputs-12 outputs Audio processor

UAPG2-8In8Out

8 inputs-8 outputs Audio processor

UAPG2-12In4Out

12 inputs-4 outputs Audio processor

UAPG2-16In

16 inputs Audio processor

UAPG2-16Out

16 outputs Audio processor

Designed for Commercial applications, the UAPG2 is the new DSP expandable universal audio processor for medium paging and multizone audio routing applications. With its powerful audio digital signal processing, the UAPG2 can easily be used in a demanding environment for high audio quality. Using the Ethernet port, it is easy to connect and control the UAPG2 through an IP network or direct from a PC.

Sound quality

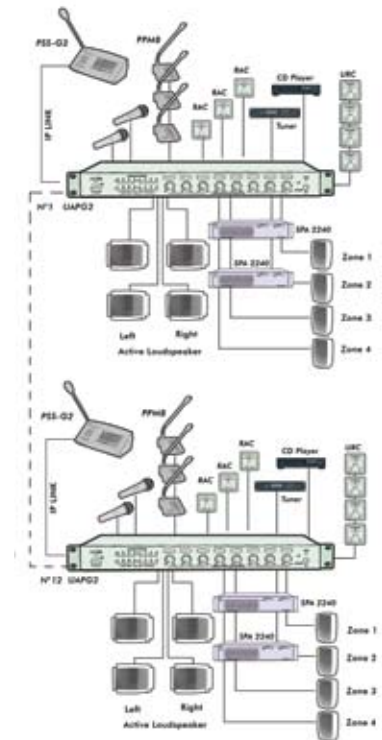
The advanced 24 bit A/D and D/A converters, together with the 48/96 kHz capable audio processing and the ADSP 21371 DSP (266 MHz SIMD SHARC Core, capable of 1596 MFLOPS peak performance), guarantee an excellent sound quality with the lowest possible latency.

Easy to use PC software for system design and control (GUI)

The UAPG2 System software provides all the necessary tools to set up and control the hardware and software elements of the UAPG2.

Impressive library of signal processing tools

The UAPG2 is a comprehensive system which integrates pre-amplifier, compressor-limiter and equalizer, as well as



matrix and delay functions into one unit. Useful features like Automatic Gain Control, Feedback killers, Automatic Microphone mixers and Crossovers, Automatic Noise sensing, are also part of the UAPG2 DSP component library. Internal processing of audio signals can be fully programmed to suit the client's application. Installers can select the audio processing feature(s) that they wish to apply to the various inputs and outputs from a library on their PC, using software provided with the UAPG2.

Once the configuration process is completed, it can be loaded into the UAPG2. All configurations can be backed-up onto PC and loaded into the UAPG2 as and when required.

UAP System

UAPG2

Expandable Universal Audio Processor



Advanced Preset manager

The UAPG2 includes two types of presets:
 More than 16 Parameter presets: They enable values of multiple parameters of the same design, such as levels, gains, EQ, etc. to be restored either from the PC software, the remote controllers or the control inputs.
 More than 32 Design presets: They enable completely different designs to be restored. Application examples for this feature are hotel meeting rooms with removable walls.

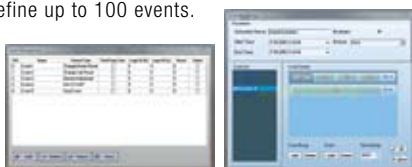
Message player

The Message player incorporated into the UAPG2 allows you to play any kind of message to be played. Only one message per UAPG2 can be run at a time. With the 100 Mbyte memory, the following storage times are available with WAV format:
 - 53 minutes of audio message at 16 kHz, 16 bits
 - 34 minutes of audio message at 24 kHz, 16 bits
 - 17 minutes of audio message at 48 kHz, 16 bits



Scheduler and event management

The scheduler allows planning of events (preset change, message play, close/open TTL out or change component's adjustments). Up to 128 different schedules can be scheduled. In one schedule you can define up to 100 events.



Control inputs

The UAPG2 has 16 (0 to 5 VDC) control inputs either analogue or Logical (Logical). Each control can be associated to any of the variable audio processing functions of the UAPG2 (input

level, output level, equalization, routing, mute, bypass, preset change...). Several parameters (Min + Max values, positive or negative variation, linear, log, anti-log) can be programmed for each of these controls.

8 Logic outputs (GPIOs)

Each UAPG2 is equipped with 8 logic outputs (common rail contact). Each of those hardware outputs can be associated to virtually any software buttons or LEDs the system designer requires. The logic outputs can be used to enable the UAPG2 to control external equipment.

RS-232 serial interfacing for third party control

The UAPG2 can be controlled from third party equipment like Vity, AMX or AMX, Creston or Vity roomcontroller via its RS232 serial port via its RS232 serial port, or IP network.

Ethernet port

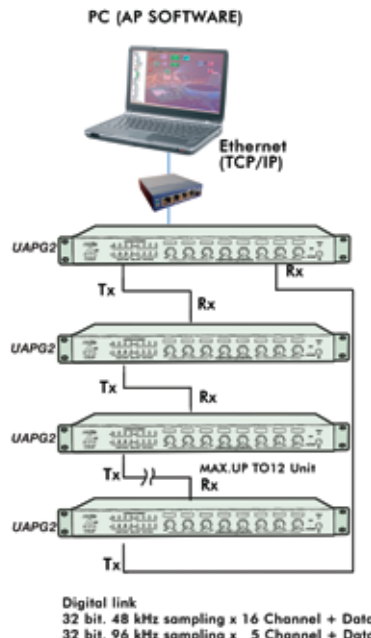
The UAPG2 can be programmed, controlled and also monitored via IP network using its RJ45 connector.

Junction Box

Easy chain-connection of UAPG2 peripherals (URC and PPM WJB-V3), using standard CAT5 cables. Junction box is included with the PPMWJB-V3.

UAPG2 Digital Link

If more inputs or outputs are required, it is possible to digitally link up to 12 UAPG2 (maximum distance between two UAPG2 is 10 meters). Through this link, you can share up to 16 channels at 48 kHz or 5 channels at 96 kHz sampling rate to the next device.



UAPG2 CHARACTERISTICS

AUDIO INPUTS

- Audio input impedance: 10 kOhms (symmetrical, screw terminal).
- Input sensitivity: 0 dB, -12dB, -24 dB, -40 dB, -55 dB (software selection).
- Max input: +15 dBu.
- Bandwidth: 20 Hz to 20 kHz.
- Phantom power soft config 48 VDC.

AUDIO OUTPUTS

- Audio output impedance: 100 Ohms (symmetrical, screw terminal).
- Bandwidth 20 Hz to 20 kHz
- Max output +15 dBu.
- Total Harmonic Distortion < 0.03% , +0 dBu, 20~20 kHz, unity gain, 20 kHz BW.
- S/N: (100 dB),(re+15 dBu), unity gain, 22 kHz BW.
- S/N:(80 dB),(re+15 dBu),(54 dB gain), 22 kHz BW.

SERIAL CONNECTIONS

- RS232 port: for ATEIS or third party equipment remote control.
- RS485 port: for Remote Controllers and Paging Microphones data control.
- RJ45 port: for PC control and system set up.

DIGITAL LINK

- One RJ45 TX to send data and audio to next device.
- One RJ45 Rx to receive data and audio from the previous device.

CONTROL OUTPUTS

- 8 common rail contact outputs. Comment : Caution ! you cannot choose between NO and NC.

CONTROL INPUTS

- 16 Logical inputs or analogue inputs: 0 to 5 VDC (software selection).

SIZE AND UNIT

- Metal unit 1 U 19" grey RAL 7016.
- L x W x D: 431 x 44 x 240 mm.

POWER SUPPLY/CONSUMPTION

- Power supply: AC: 100 to 240 VAC 50/60 Hz DC: 21-28 VDC.
- Consumption: 40 VA
10 VA in stand-by mode.

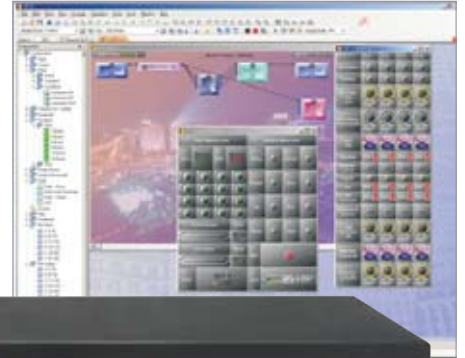


ECS

Networked Echo Cancellation System



ATEIS-NET graphical user interface



ECS Features

- Dynamic Echo Cancellation tail time; Shifted - Standard 130 ms, max. 260 ms
- Noise reduction: 6 – 20 dB.
- Up to 16 AEC software modules.
- Up to 24 input / output channels.
- Powerful 48 channel Networking.
- ATEIS-NET ports with up to 32 multi-unit system designs.
- Powerful networked automixer, for up to 256 delegates within one system.
- Programmable logic in-and outputs.
- Ethernet port for remote diagnostic and system configuration.
- Support IP telephone with SIP protocol and G.711 G.722 voice codec via Ethernet port.
- Up to 16 broad-band AEC software modules.
- From 4 up to 24 balanced mic/line inputs/outputs combinations (See type number specification).
- Serial port for third-party RS-232 remote control.
- Remote control bus for dedicated control panels.
- ATEIS-STUDIO configuration software for Windows®.
- Modular I/O with pre-definable processing.
- RoHS compliance, CE marked RoHS compliance, CE marked.
- Ability to select, view, and adjust within ATEIS-STUDIO graphical user interface:
 - Mixers: standard, automatic, matrix, combiners
 - Equalizers: graphic, parametric, feedback
 - Filters: HPF, LPF, high shelf, low shelf, all-pass
 - Crossovers: 2-Way, 3-Way and 4-way
 - Dynamics: leveler, comp/limiter, ducker, ANC
 - Routers: 2x1 ~ 768x1 and 384x384
 - Delays: 0 ~ 2000 ms
 - Controls: levels, presets, logic, RS-232, etc.
 - Meters: signal present, peak, RMS
 - Generators: tone, pink-noise, white-noise
 - 2 years full warranty.

ECS

- ECS 111**
4 inputs/4 outputs/4 AEC-channels
- ECS 211**
8 inputs/4 outputs/4 AEC-channels
- ECS 212**
8 inputs/4 outputs/8 AEC-channels
- ECS 222**
8 inputs/8 outputs/8 AEC-channels
- ECS 322**
12 inputs/8 outputs/12 AEC-channels
- ECS 413**
16 inputs/4 outputs/12 AEC-channels
- ECS 514**
20 inputs/4 outputs/16 AEC-channels
- ECS T**
Optional Telephone hybrid card

The ATEIS Networked Echo Cancellation System, is a digital signal processor featuring DYNAMIC Automatic Echo Cancellation based on our own developed wideband acoustic echo cancellation algorithm. Designed specifically to provide clear audio in teleconferencing applications. The ECS delivers true 20 Hz to 22 kHz bandwidth during multiple participant conversations, with natural full duplex transmission of speech with no latency, including double talk.

Driven by the ATEIS STUDIO graphical user interface, the ECS includes a broad selection of audio components, routing options and signal processing. The ECS is a comprehensive system which integrates compressor-limiter and equalizer, as well as matrix and delay functions into one unit. Useful features like Automatic Gain Control, Feedback killers, Meters, Generators and Automatic Microphone mixers are also part of the ECS DSP component library. Internal processing of audio signals can be fully programmed to suit the client's application. Installers can select the audio processing feature(s) that they wish to apply to the various inputs and outputs from a library on their PC, using software provided with the ECS. The internal system design is completely user definable via PC software, and can be controlled via dedicated software screens, RS-232 control systems, and a variety of optional remote controls.

Multi-unit ECS-systems can be created utilizing Ethernet and ATEIS NetLink light.

Telephone hybrid capabilities include: initiation of outgoing calls detection and answering of incoming calls.

ECS features up to 16 wide-band AEC software modules and up to 24 configurable hardware electronically balanced inputs and/or outputs, telephone interface with line & set connections, Ethernet port for software configuration/control, serial port for third-party RS-232 remote control and remote control bus for dedicated control panels like the URC-programmable remote controllers.

ATEIS NetLink light provides 48-channels broadband audio distribution, linking up to 32 units in a dedicated network creating a through 768 channels audio-conferencing system.

ECS System

ECS

Networked Echo Cancellation System



ECS CHARACTERISTICS

AUDIO INPUTS

- Audio input impedance: 10 kOhms (symmetrical, screw terminal).
- Input sensitivity: 0 dB, -12dB, -24 dB,-40dB, -55 dB (software selection).
- Max input: +15 dBu.
- Bandwidth: 20 Hz to 20 kHz.
- Phantom power soft config 48 VDC.

AUDIO OUTPUTS

- Audio output impedance: 100 Ohms (symmetrical, screw terminal).
- Max output +15 dBu.
- Bandwidth 20 Hz to 20 kHz.
- Total Harmonic Distortion < 0.03% , +0 dBu, 20~20 kHz, unity gain, 20 kHz BW.
- S/N: (100 dB),(re+15 dBu), unity gain, 22 kHz BW.
- S/N:(80 dB),(re+15 dBu),(54 dB gain), 22 kHz BW.
- Dynamic range: (100 dB),(re+15 dBu), 22 kHz BW.

SERIAL CONNECTIONS

- RS232 port: for ATEiS or third party equipment remote control.
- RS485 port: for Remote data control.
- RJ45 port: for PC control and system set up using ATEiS-STUDIO software.

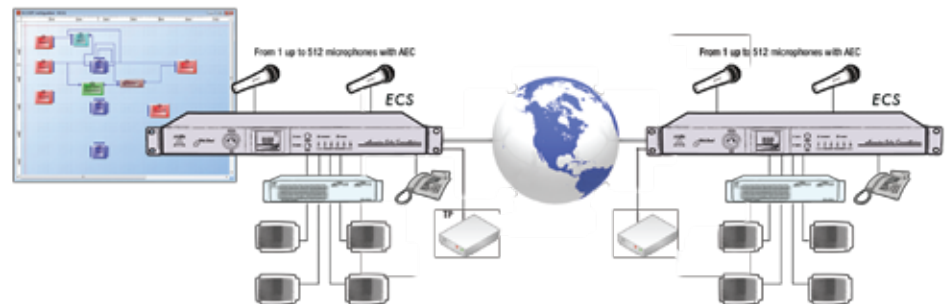
DIGITAL LINK

- One RJ45 TX to send data and audio to next device.
- One RJ45 Rx to receive data and audio from the previous device.

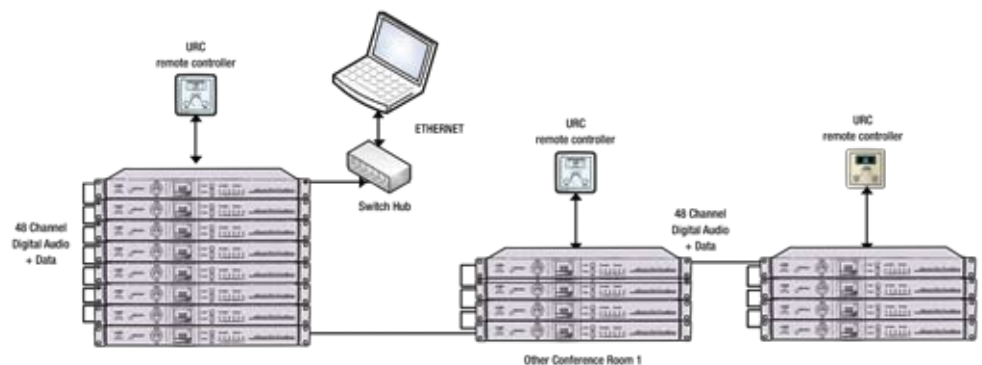
CONTROL OUTPUTS

- 8 common rail contact outputs.

Audio Telecommunication



Networked Echo Cancellation System





RAC5 / RAC8 / URC / URC200 / NSM

Controller Devices

RAC5 / RAC8 /

5 / 8 Steps remote controller.

Wall mounted level and 5 or 8 sources selectors.



URC / programmable remote controller (RS485).

Programmable remote controller with display. Furthermore, custom control panels can run on a PC connected to the UAPG2 network via the RJ45 connector or you can use 3rd party protocol control via RS232.



URC200 / Programmable remote controller (TCP/IP).

The URC200 is an programmable remote controller (TCP/IP) for the LAPG2 & ECS DSP audio matrix System and the IP-media streamers with VOX-NET IP Media software. The URC200 is powered over IP and easy to integrate with current demands for room controllers like light, curtains, sound and video control.

The full color display is easy to read and has a low-power consumption to allow for long lines and multiple devices into one system.

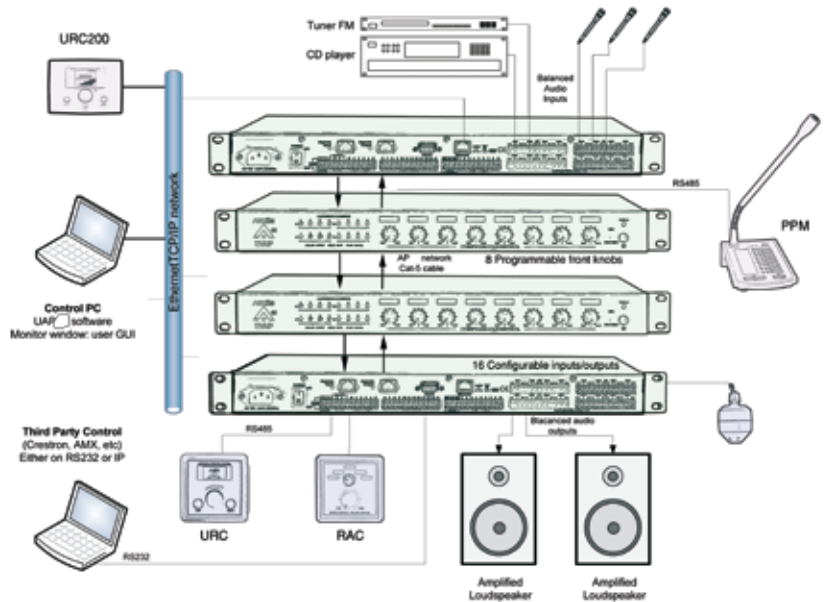


URC200 CHARACTERISTICS

- Screen size: 2' FULL COLOUR
- Function keys: 2
- Rotary switch: 1
- Dimensions: 140x107x34 (28 flush mounted) WxHxD
- Power: 24 VDC or POE
- Features: IR-receiver
 Ethernet control
 VOX-NET IP-media software

NSM / Sensing microphone

The NSM assures the omnidirectional sound recording and preamplification of the surrounding background noise. The 0 dB modulation is sent through the UAPG2 / LAPG2 in order to provide the automatic gain control feature of the A.N.G (Auto Noise Gain) component and allows the level adjustment where the NSM is implemented.



Typical example with UAPG2 DSP audio matrix System

COMPATIBILITY	UAPG2	LAPG2	ECS
PM1	V Using one audio input + one contact. Limited to the amount of audio-inputs per system.	V Using one audio input + one contact. Limited to the amount of audio-inputs per system.	V Using one audio input + one contact. Limited to the amount of audio-inputs per system.
PPMWJB-V3 & PPM Keypad G2	V Max. 16 on an unit using 16 audio inputs. Max 32 in a system sharing the same RS485.	V Max. 16 on an unit using 16 audio inputs. Max 32 in a system sharing the same RS485.	I Max. 16 on an unit using 16 audio inputs. Max 32 in a system sharing the same RS485.
PPMIT5	X	V Max. 1 pcs over TCP/IP using 1 input	V Max. 1 pcs over TCP/IP using 1 input
NSM	V Applicable on any input. Max number limited to DSP system occupation.	V Applicable on any input. Max number limited to DSP system occupation.	V Applicable on any input. Max number limited to DSP system occupation.
RAC5	V Max 16 per device	V Max 16 per device	V Max 8 per device
RAC8	V Max 16 per device	V Max 16 per device	V Max 8 per device
URC	V Max 8 per device	V Max 8 per device	V Max 8 per device
URC-200	I TCP/IP - max 256 in a system	I TCP/IP - max 256 in a system	I TCP/IP - max 256 in a system

I: INFO; ask for availability

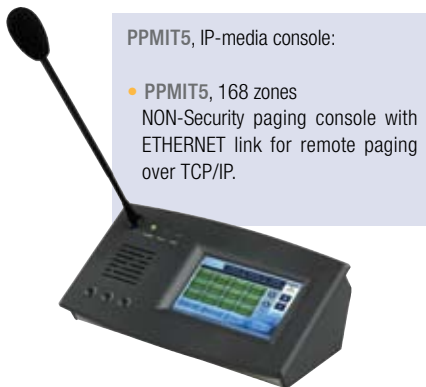
Consoles & Accessories

PPMIT5 / PPMWJB-V3xx / PM1

Paging Consoles

PPMIT5 Generic information

Paging console for, LAPG2 and ECS System.



PPMIT5, IP-media console:

- PPMIT5, 168 zones NON-Security paging console with ETHERNET link for remote paging over TCP/IP.

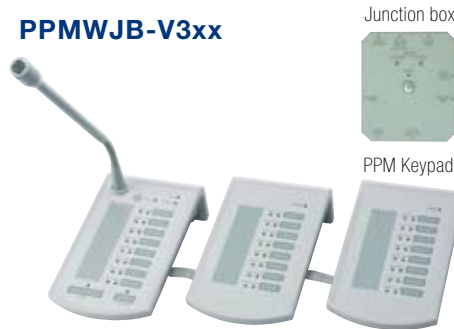
The PPMIT5 IP media console is a man-machine interface which allows call-paging, messages broadcasting and DSP matrix parameter control. Its back-lit touch screen is designed for simple and user-friendly operating. The 3-hardware-keys can be freely assigned within the System control software. Various operating levels with password protection make the PPMIT5 Media console a versatile device that fits well in a commercial shopping center as for an industrial environment where paging over IP-networking brings flexibility and easy access.

All paging parameters needed for site operating can be programmed: zones assigned to the different buttons, name of zones, group of zones, messages triggering, levels adjustments and pre-call chime but also for fader control, button control or event control.

The message and the chime are stored into the PPMIT5 IP media console.

A total of 168 keys over 12 pages allow zone or group of zones selections. Each key contains a green colour changing field indicating that the zone is occupied by a different process.

PPMWJB-V3xx



Unidirectional Condenser Addressable Microphone, compatible with all system units, PPMWJB-V3xx uses an RS485 protocol over a single CAT5 cable connection, to transport both Audio and Power from the PPMWJB-V3xx to the system units.

The PPMWJB-V3xx comprises of 8 zones / 8 buttons with sleek condenser gooseneck microphone, and spring metal protection, providing durability and excellent aesthetics as well as allowing up to 256 zones expansion via the additional Keypad easy extension station. The buttons can represent a single zone or a group of zones and are easily defined via the GUI of the system units using a simple Matrix selection.

The unit offers "Hold" and "Busy" LED signals in addition to the zone LED's, and these allow the easy identification of selection / Busy signals for the user.

All buttons can be programmed with drag & drop features from the System unit GUI software and each button can be programmed for Push To Talk or Latch functionality. The unidirectional condenser microphone warrants high quality directive signal pick up from the user and hence less interference from the surroundings thanks to the cardioid polar pick-up pattern.

The RS485 communication protocol offers daisy chaining of up to 100 m on a CAT5 cable, and yet makes outlets easy to connect via a standard RJ45 connectors. (The microphone compatibility listing shows the maximum number of units per System).

Main Features: - 8 paging programmable keys with status and 1 All call button

PPMIT5 CHARACTERISTICS

- 5" TFT full colour paging console
- 3 LED indicators: Power/General Fault/Evacuation active
- Ethernet: CAT5 connection
- 3 key-buttons: User definable using ATEIS Studio GUI
- 168 Touch fields: 14 pages with 12 keys ,
- Console size: (L x W x D) 250 x 140 x 80 mm
- Microphone flex length: 280 mm
- Weight: 1098 g
- Colour: RAL 7016
- Front tilted at 30°
- Material: metal back, PVC top and sides

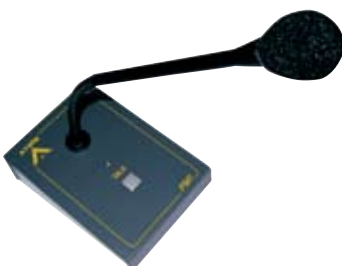
PPMWJB-V3xx CHARACTERISTICS

- Base dimensions (HxWxD): 200 x 107 x 345 mm
- Weight: Approx.0.55 kg
- Colour: Ral 7035
- Gooze-neck length with mic.: 300 mm
- Cable length: 900 m
- Comes standard with Junctionbox (JB) and CAT5 cable (1.5 m).
- PPM Keypad G2 optional Expansion keypad for an additional 8 selection zones

PM1 CHARACTERISTICS

- Height: 45 mm
- Length: 250 mm with flex 90 mm
- Dimensions: 130 x 90 mm
- Height: 45 mm
- Colour: Ral 7035
- Red / Green Bi-colored LED:
Green: powered - Red: line busy
- Entry contact: toggle the LED in red

PM1 / Paging console



The PM1 is a preamplified single button desktop paging console that links on all the products, UAPG2, LAPG2 and ECS. This desktop paging console with PTT is meant for a simple All-Call or fixed designated area calls.

It comes with a robust gooseneck microphone and has a symmetric line-level raised output. (0 dB).

The unit is 24 VDC powered. Optional PSU available.

Functionality:

- Desktop powered ON: permanent green LED
- Contact Out: activation on call key (PTT)
- Contact In: to light the red LED (line busy indicator)

Public Address - Voice Alarm

IPVA - Intercom

Commercial Audio

Intelligent Acoustic Solutions

Loudspeakers



ATEiS Europe B.V.

Sydneystraat 42 - 3047 BP - ROTTERDAM - NETHERLANDS

Phone +31 (0)10 2088690 - Fax +31 (0)10 2088699

www.ateis-europe.com

info@ateis-europe.com

DELIVERING YOUR MESSAGE

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