Dear Friends,

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, ATEÏS 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 ATEÏS-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 ATEÏS 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 ATEÏS is

**Delivering Your Message!**
Delivering Your Message

Commercial Audio

LAP DSP audio matrix System

Designed for large sized prosound 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.

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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 multi-zone audio routing applications.

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UAPG2 / Expandable Universal Audio Processor

ECS 24 I/O Drag-N-Drop DSP with AEC

The ECS is a drag-n-drop digital signal processor that can be used in a multitude of environments and also features optional dynamic echo cancellation. The AEC chip is based on our in-house developed wideband acoustic and line cancellation algorithm designed specifically to provide clear audio in teleconferencing applications.

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ECS / 24 I/O Drag-N-Drop DSP with AEC

Consoles and Accessories

Designed for large sized prosound 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.

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PPMIT5 / PPMWJB-V3xx / PM1 / Paging Consoles
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 able to store several audio message in the LAPG2 that can play 2 messages at the same time and store up to 53 minutes of text/music. (is not available in 96 kHz mode).

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 a control inputs 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.

- Up to 32 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.
16 Analogue control or logic inputs

The LAPG2 has 16 control inputs which can be configured as analog control input (0 to 5 VDC) or logic input (TTL). Each analog 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. The logic inputs could be used to mute or activate an audio signal. The logic inputs can be used in normal or binary mode.

8 Logic outputs (GPIOs)

Each LAPG2 is equipped with 8 relay outputs. Each of those outputs can be associated to virtually any software button the system designer requires. The logic outputs enable the LAPG2 to control external equipment.

RS-232 serial interfacing for third party control

The LAPG2 can be controlled from third party equipment like Vby, 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 LAPG2.

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 132 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.

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
- EWAC: red
- Stand by: green
- Data: green
- Ethernet: green
- ATEIS-NET: green
The UAPG2 features 4 slots that can be populated with either input or an output card. The input cards feature 4 channels of mic or line level inputs and the output card for channels of line level output. Optional AES/EBU cards are also available.

The UAPG2 features 8 programmable and scalable knobs on the front. These knobs can be programmed to control any function in the software library and scaled to suit the application.

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.
**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.
- UP to 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:

- 36 minutes of audio message at 48 kHz, 8 bits
- 18 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 (TTL). 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, antilog) can be programmed for each of these controls.

**16 Logic Inputs and 8 Logic outputs (GPIOs)**

Each UAPG2 features 16 logic inputs to trigger sub presets or other logic functions in the software. 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, Crestron 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 32.81 ft). 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).
- 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.
- S/N: (84 dB), unity gain 84 dB, 22 Hz/22 kHz.
- S/N: (116 dB), (54 dB gain), 22 Hz/22 kHz.

**SERIAL CONNECTIONS**

- RS232 port: for ATEÏS or third party equipment remote control.
- RS485 port: for Remote Controllers and Paging Microphones data control.
- RJ45 Ethernet 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: 16.97 x 1.73 x 9.45 in.

**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.
The ECS frame contains 6 slots that can be populated with either an input or an output card. The input cards, featuring mic preamps, contains 4 inputs and the output cards for line level outputs. If the system is used for acoustic echo cancellation in a teleconferencing application up to 4 echo cancellation chips can be added internal to the box. Each chip contains 4 echo cancellation lines for a total of 16 echo cancellation channels.

An optional telephone card can also be added for use with analog lines. In a VoIP system the VoIP is internal to the software and does not require an additional card.

The ECS Networked Echo Cancellation System, is a digital signal processor featuring an impressive DYNAMIC Automatic Echo Cancellation based on our own developed wideband acoustic echo cancellation algorithm. ECS is designed specifically to provide outstanding clear audio in teleconferencing applications. The ECS delivers true 20 Hz to 22 kHz bandwidth during multiple participant conversations, including double talk.

Special designed features like Local Echo Suppressor, Noise reduction, Remain Echo Suppressor and Voice Gate greatly improve the audio quality of your teleconferencing system.

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 (RS485) for dedicated control panels like the URC-programmable remote controllers.

Driven by the intuitive ATEÏS STUDIO graphical user interface, the ECS includes a broad selection of audio components, routing options and signal processing. Useful features like Automatic Gain Control, Feedback killers, Meters, Compressor-limiter, Equalizer, Matrix 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 with enhanced management. 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, third party devices like Crestron ™, AMX ™ or Vity ™, and a variety of optional remote controls.

Multi-unit ECS-systems can be created utilizing ATEÏS NetLink which provides 48-channels broadband audio distribution, linking up to 32 units in a dedicated network creating a through 768 channels audio-conferencing system.

The ECS system can use either telephone lines with the telephone interface or internet links thanks to a reliable VoIP stream with SIP protocol.

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

ECS

24 I/O Drag-N-Drop DSP with AEC

Audio Telecommunication

- Audio input impedance: 10 kΩ (balanced, screw terminal).
- Max input: +15 dBu.
- Bandwidth: 20 Hz to 20 kHz.
- Phantom power soft config 48 VDC.

Audio Outputs

- Audio output impedance: 100 kΩ (balanced, screw terminal).
- Max output: +15 dBu.
- Bandwidth: 20 Hz to 20 kHz.
- Total Harmonic Distortion < 0.03%, +15 dBu, 20–20 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 I/O

- 8 Control inputs either analog or Logical (contacts).
- 2 logic outputs (Dry contacts).
RAC5 / RAC8 / URC / URC200 / NSM

Controller Devices

RAC5 / RAC8 / URC200

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.

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.

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.

RAC5 / RAC8

Typical example with UAPG2 DSP audio matrix System

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I: INFO, ask for availability

URC200 CHARACTERISTICS

- Screen size: 2’ FULL COLOUR
- Function keys: 2
- Rotary switch: 1
- Dimensions: 5.51 x 4.21 x 1.33 (28 flush mounted) WxHxD
- Power: 24 VDC or POE
- Features: IR-receiver
  Ethernet control
  VOX-NET IP-media software
PPMIT5 / PPMWJB-V3xx / PM1

**PPMIT5 Generic Information**

Paging console for, LAPG2 and ECS System.

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, name of zones, group of zones, messages triggering, levels programmed: zones assigned to the different buttons, 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 digital 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 984.252 yd 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

**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 goose neck 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 digital 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 984.252 yd 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

**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)

**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) 9.84 x 5.51 x 3.15 in.
- Microphone flex length: 11.02 in.
- Weight: 2.42 lbs.
- Colour: RAL 7016
- Front tilted at 30°
- Material: metal back, PVC top and sides

**PPMWJB-V3xx Characteristics**

- Base dimensions (HxWxD): 7.87 x 4.21 x 13.58 in.
- Weight: Approx.1.2 lbs.
- Colour: Ral 7035
- Gooze-neck length with mic.: 11.81 in.
- Cable length: 2.952.76 ft.
- Comes standard with Junctionbox (JB) and CAT5 cable (4.92 ft.).
- PPM Keypad G2 optional Expansion keypad for an additional 8 selection zones

**PM1 Characteristics**

- Height: 1.77 in.
- Length: 9.84 in. with flex 3.54 in.
- Dimensions: 5.12 x 3.54 in.
- Height: 1.77 in.
- Colour: Ral 7035
- Red / Green Bi-colored LED: Green: powered - Red: line busy
- Entry contact: toggle the LED in red
Public Address - Voice Alarm
Audio Distribution over IP
Commercial Audio
Intelligent Acoustic Solutions
Intercommunication
Loudspeakers

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