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New Public Address & Voice Evacuation System

The Smart approach to better understanding

smartVES

SmartVES is a new PAVA system utilising innovative technologies that improve the intelligibility of broadcast voice messages in buildings with difficult acoustic conditions.

In addition to the typical components for powering, battery charging, routing, amplification and transmission of messages to individual public address zones, the system is equipped with an all-new SMART-DU-1604 central unit, responsible for **monitoring acoustic conditions** in the building and **modifying the broadcast voice signal** both in timing and frequency in order to obtain the maximum possible intelligibility of the transmitted messages.

The modifications to the audio signal are performed by powerful digital signal processing in a real-time mode utilising continuous acoustic feedback into the processors to apply adaptive filtering. This feedback to the processors of the live audio performance in the audio zone is realised through specially designed SMART-ANSM-01 measuring microphones installed in the listening area. The collected signals are transmitted via the SMART-AMAP-6 microphone aggregation point to the SMART DU-1604 central units.

Each hub can collect measurement signals from up to 6 SMART-ANSM-01 microphones and transmit them to the SMART-DU-1604 via the LAN network.

Tha advantages of smartVES technology:

» regardless of the acoustic conditions on the site, guarantees the required SPL level and required SNR level in each frequency band of the transmitted messages to intelligently maximise intelligibility;

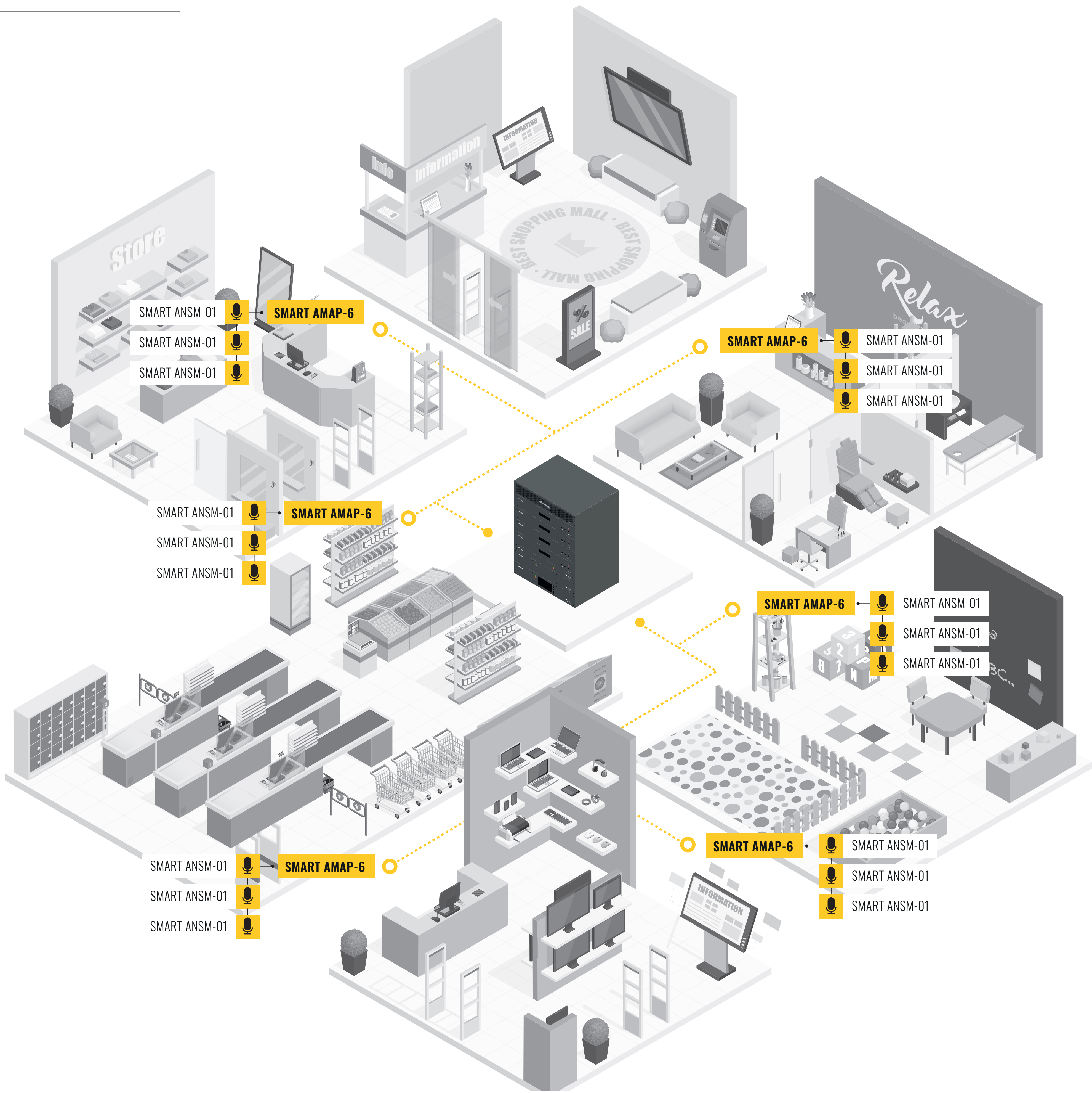
» can remove the need to install additional acoustic treatments in a building to achieve intelligible announcements;

» simplicity of configuration;

» quick start-up, thanks to the auto-calibration function, the system detects the positions of the SMART-ANSM-01 measuring microphones in relation to the loudspeaker zones;

» automatic, multi-point process of adjusting the sound of the loudspeakers and their levels to the changing conditions in the listening area;

» a unique algorithm that changes the pace and length of speech in real time, maximizing the intelligibility of the transmitted messages.



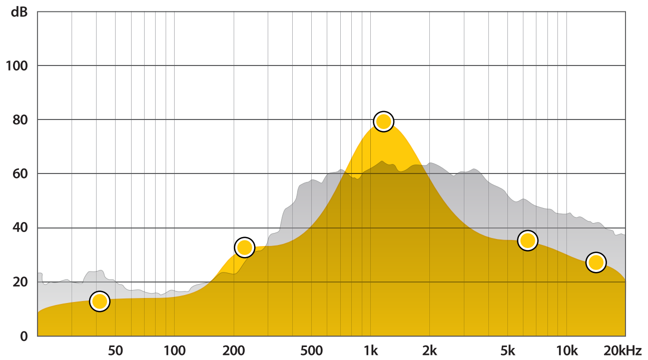
smartVES is a **unique solution** in that, regardless of the prevailing acoustic conditions in the facility, **smartVES maximizes the speech intelligibility parameter dynamically**, using the full technical capabilities of speakers, amplifiers and the settings of signal processors. The advanced method of auto-calibration and of mapping the matrix of SMART - ANSM-01 measuring microphones to the loudspeaker zones makes the complete system **extremely simple and quick to configure** with most of the settings being selected automatically.

The adaptive filtration (AF) algorithm

smartVES ensures the maximization of the STI coefficient and maintains the appropriate headroom (5, 10 or 15 dB) between the useful signals, e.g. spoken messages, and any undesirable acoustic background.

This process of the maximisation of the STI is calculated by a number of innovative algorithms implemented in the SMART-DU-1604 unit. The key algorithms include the algorithm for adaptive filtering and temporal transposition of the speech signal, as well as algorithms for calculating SNR, STI and auto-calibration of the system, supporting real-time measurements, whose task is to obtain information that allows optimal adjustment of the sound to the current acoustic conditions.

The adaptive filtration (AF) algorithm works automatically. The main task of this algorithm is to modify the shape of the correcting function on an ongoing basis, depending on the acoustic conditions without the participation of the operator. The algorithm changes the frequency response and amplification of the sound reproduced by the loudspeakers based on the impulse response of the room, the current spectrum and noise level as well as the maximum and minimum SPL level generated by the system in a given zone. A single SMART-DU1604 processor can simultaneously process up to 16 audio streams, corresponding to 16 speaker zones that are subject to independent adaptive filtering.

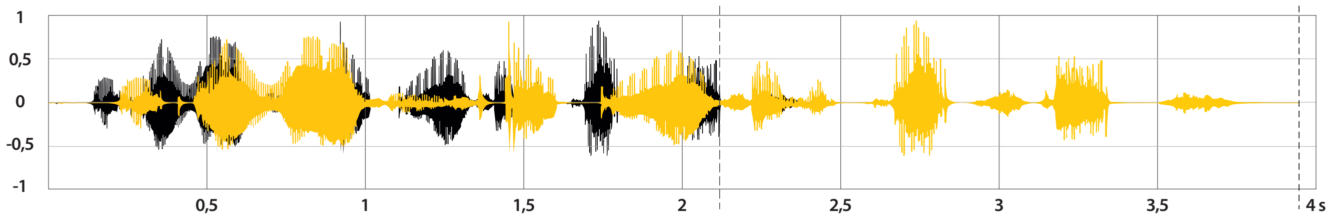


The speech temporal transposition algorithm (STTA)

The speech temporal transposition algorithm (STTA) naturally and evenly changes the pace and duration of messages spoken in real time through smartVES system microphones (DFMS, DMS, DMS-LCD). The algorithm can distinguish the type of voice (male / female), determines the rate of speech, and most importantly, detects and shortens the duration of stuttering, while stuttering is understood as excessive prolongation of the articulation of the selected sound.

The first processing stage is carrying out the voice detection procedure. When a speech signal is detected, the system determines its rate while buffering the signal. Then, based on the value of the STI index measured in the auto-calibration process, the instantaneous value of the deceleration factor of the speech signal is determined.

As a result, the output of the STTA algorithm obtains a natural speech sound, the pace of which is adjusted to the acoustic conditions in the area of message broadcasting, ensuring the maximization of intelligibility, regardless of the language of the messages or the speech speed.



Example applications

Example 1: Stair case

Due to the influence of factors such as:

» long reverberation time,

» noise generated by the smoke exhaust system,

» noise of moving people,

the staircase is an acoustically difficult area, where difficulties in obtaining satisfactory speech intelligibility should be expected.

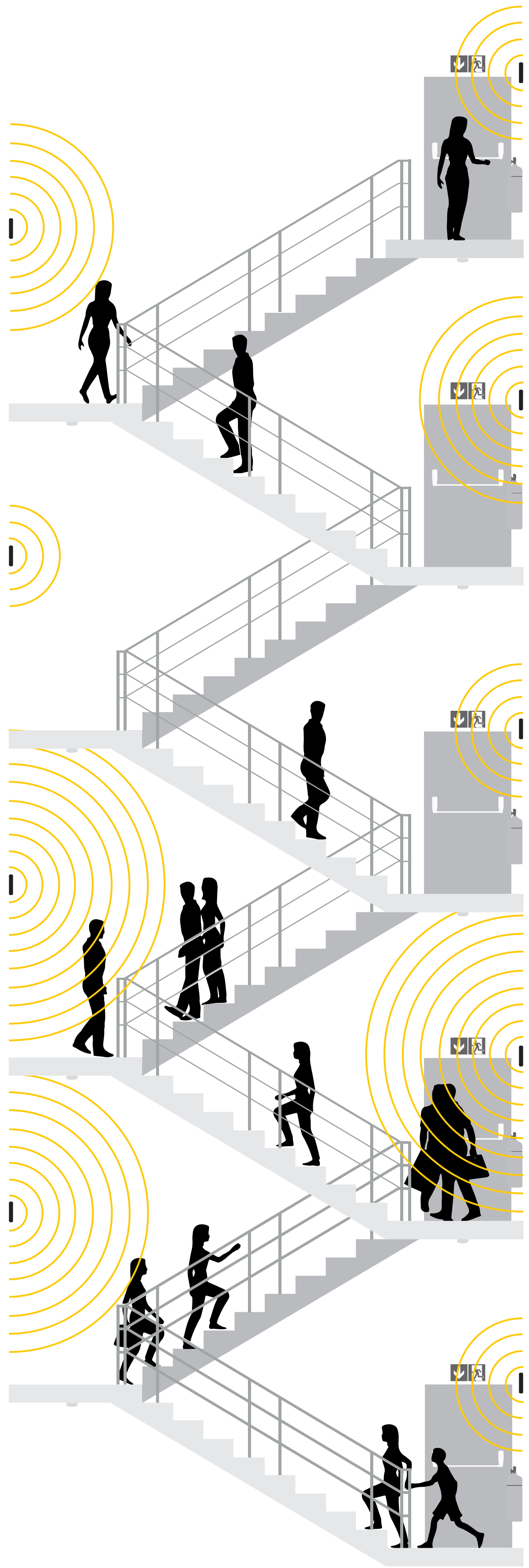
For optimal speech intelligibility, the Voice Alarm System must ensure a sufficient distance between the sound pressure level of the voice message and the background noise level. At the same time, care should be taken not to excessively stimulate the reverberation in the room.

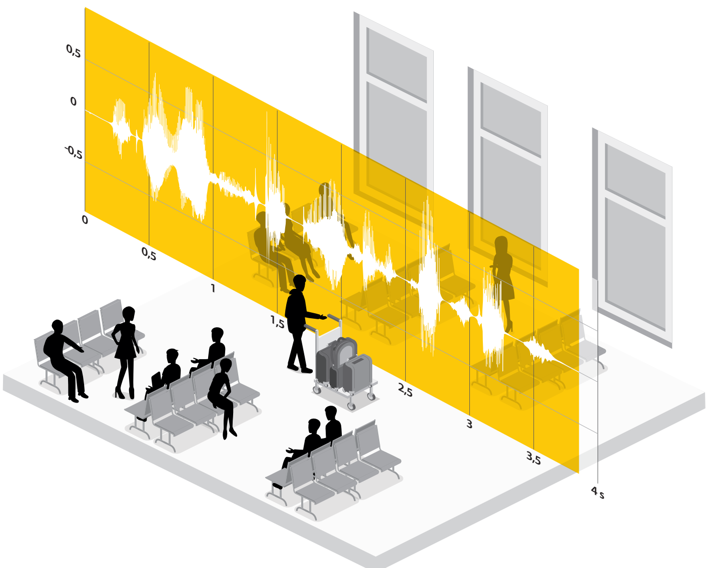
smartVES will automatically adjust the level of voice message emission in the zone, introduce a tone correction optimized for the impulse response of the room as well as the frequency characteristics of the measured noise.
smartVES reacts dynamically to changes in acoustic conditions in the room - the introduced sound color correction will depend on the actual reading of the noise characteristics in the room.

Example 2: Terminal

smartVES ensures increased efficiency of message transmission, regardless of whether these messages are played from the system memory, an external audio source or delivered in real time using system microphones.

In case of messages delivered manually by the operator, factors such as timbre, speech rate and diction will have a significant impact on speech intelligibility. In case of untrained speakers, a very frequent problem is the speed of speech. In large rooms with a long reverberation time,





a statement given too quickly will be incomprehensible.

smartVES is equipped with a unique speech temporal transposition algorithm (STTA), which allows to change the pace and length of speech in real time, thanks to which, in difficult acoustic conditions, its perception increases.

The speech temporal transposition (STTA) algorithm can simultaneously process the signals from 4 system microphones.

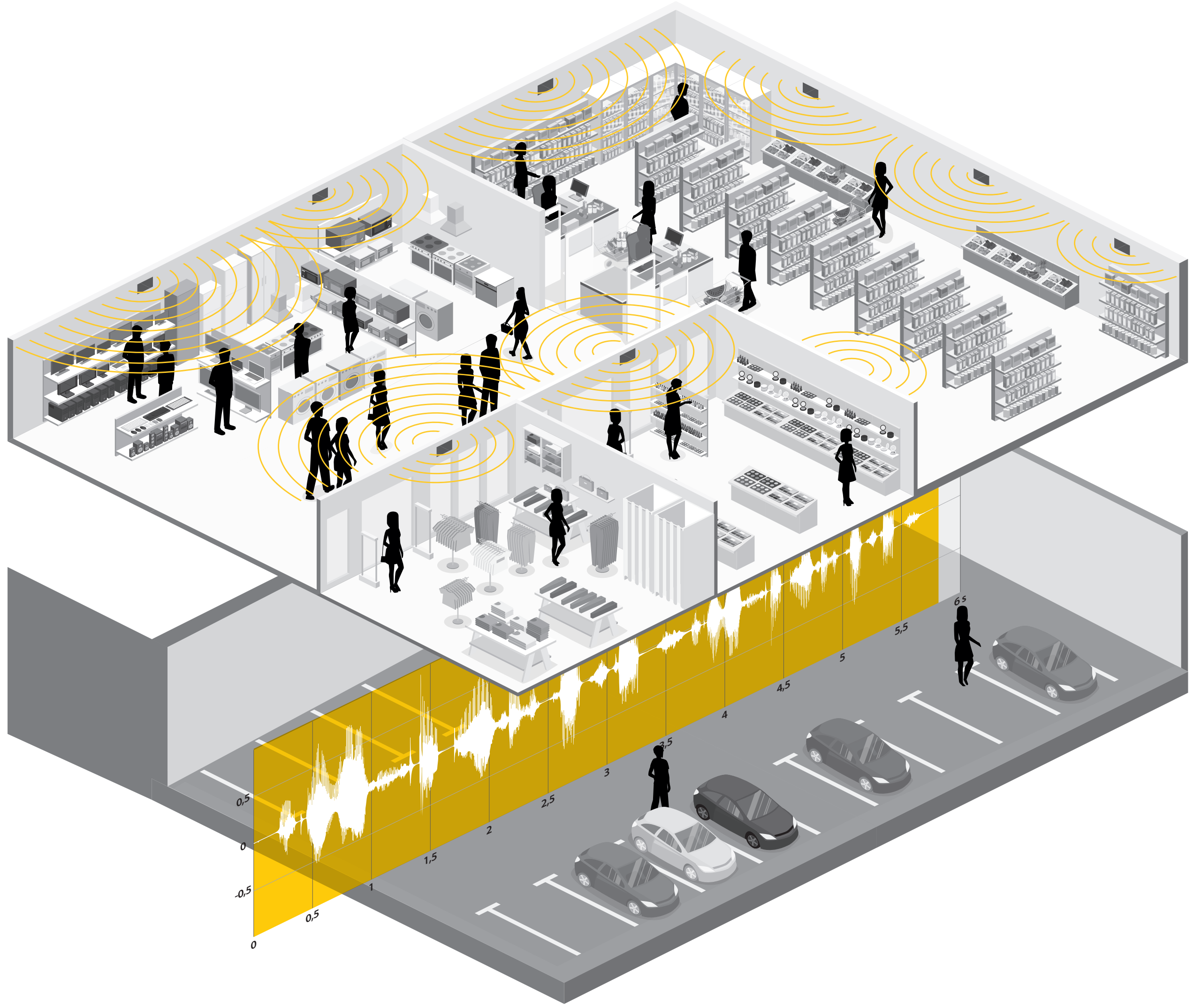
Example 3: The shopping mall

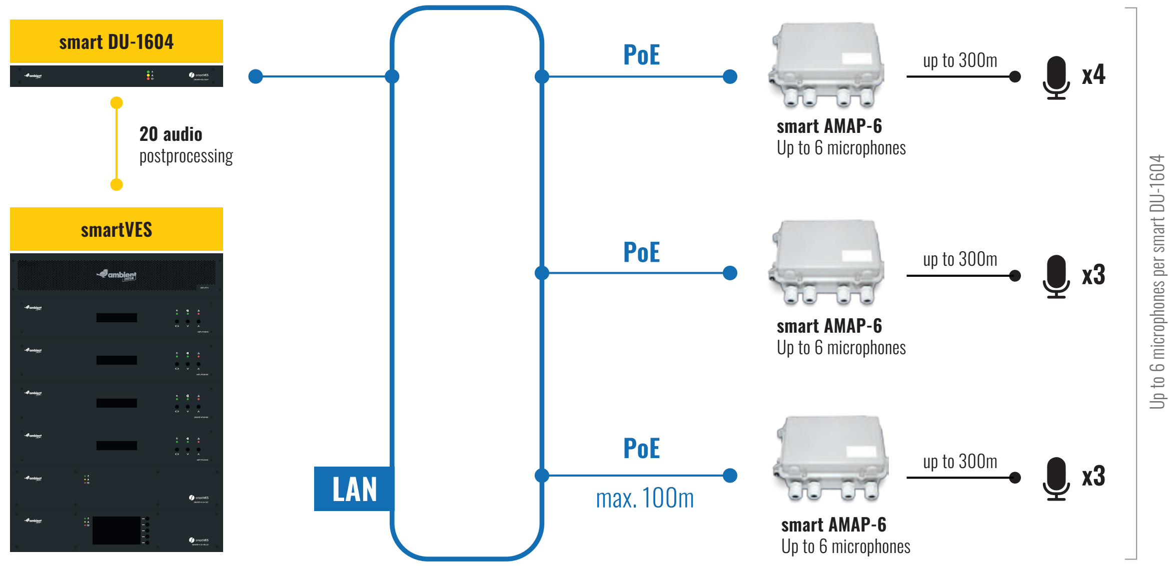
The adaptive filtering (AF) and speech temporal transposition (STTA) algorithms can work simultaneously. This allows you to maximize the effectiveness of the system.

Measurement data necessary for the correct operation of the algorithms are delivered to the SMART-DU-1604 processor from the SMART-ANSM-01 measurement microphones, which should be located in the zones with sound.

Measurement microphones are powered from SMART-AMAP-6 hubs, which communicate with the DU-1604 processor via the LAN network.

A single SMART-DU-1604 processor enables the processing of up to 16 measurement signals.





Measuring microphone aggregation point

SMART-AMAP-6

A dedicated device has been developed for smartVES, the purpose of which is to process the signals from the measurement microphones and transmit them digitally to the SMART-DU-1604 units.

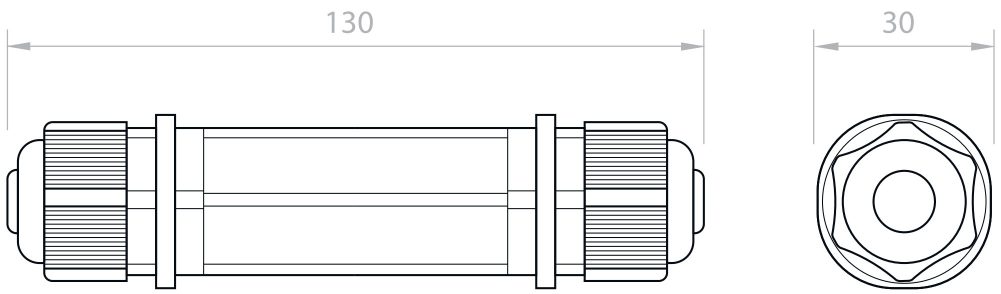
The SMART-AMAP-6 is a measurement microphone aggregation point that receives the signal from 6 measurement microphones and enables the addressing and use of their return signals in the adaptive filtering algorithms of the assigned zones.

The SMART-AMAP-6 is a networked device and can be powered via the POE of the system switch or central unit as well as from an external 40-57VDC power supply. Dedicated low-noise, microphone amplifiers available in the SMART-AMAP-6 enable the correct operation of measurement microphones up to 300m with independent monitoring circuits of each SMART-ANSM-01 individually.

Measurement microphone

SMART-ANSM-01

The measurement microphones are an integral part of the smartVES system. Their purpose is to test the level and frequency characteristics of sound at an object in the range of 100Hz-10kHz (+/- 2dB) with sound levels from 50dB to 120 dB SPL. The microphone's specially designed electronics ensure stable measurement performance with long connection cables of up to 300 metres. In addition, the dedicated housing provides easy installation and an aesthetically pleasing and unobtrusive appearance giving the microphone the ability to be used in many facilities where a small housing with multiple application possibilities is expected for functionality reasons.



About us

Ambient System is **leading Polish provider of modern PA/VA systems** to clients worldwide. Our projects range from complex installations such as **refineries, airports, stadiums, tunnels and shopping centres** to medium and small structures like **hospitals, train stations, hotels, office buildings, supermarkets or schools**.

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digital, scalable & cost-effective solutions compliant with Fire Safety industry standard EN-54

full ownership of our product cycle – design, solution development, quality testing and implementation support – all in ONE place

professional acoustic simulations and PA/VA systems design

technical expertise and specialist engineering skills

CONTACT

HEADQUARTERS

AMBIENT SYSTEM Sp. z o.o.
ul. Bysewska 27 | 80-298 Gdańsk | Poland
T: +48 58 345 51 95
ambient@ambientsystem.pl

SALES

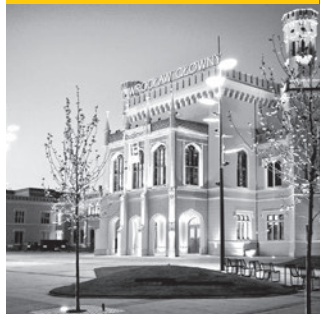
sales@ambientsystem.pl

References

Tunnel under Świna
Poland

Gdańsk Airport
Poland

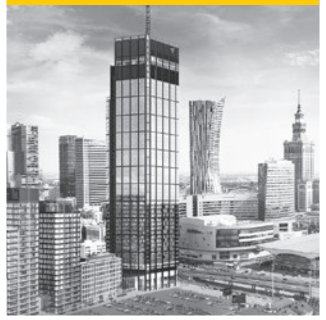
Wrocław Railway Station
Poland



Sopot Centrum Railway Station & Shopping Mall / Poland

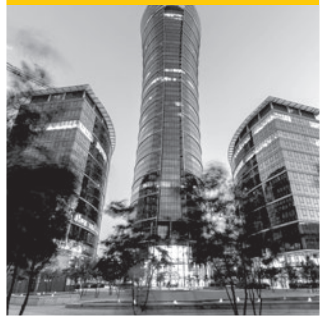
LOTOS S.A Rafinery Gdańsk
Poland

Varso Tower
Poland



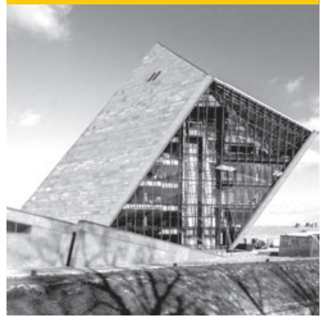
Olivia Star
Poland

Warsaw Spire Office
Poland



Office Center of Pankrac Prague
Czech Republic

Museum of the Second World War
Poland



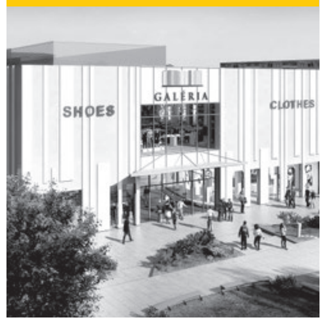
Hotel Raffles Europejski Warsaw
Poland

Bielsko-Biała Stadium
Poland

Watford Stadium
UK

Wroclavia Shopping Mall
Poland

Lučenec Shopping Mall
Slovakia

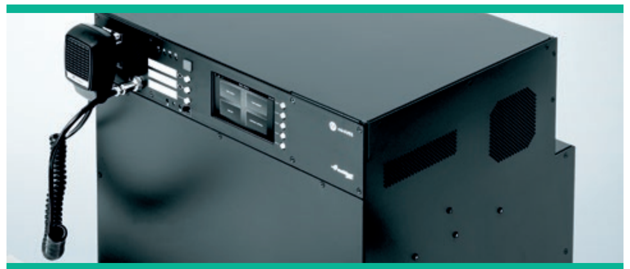


Basra Times Shopping Mall
Irak

Our products

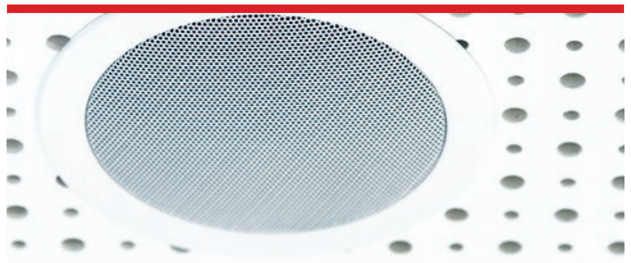
smartVES

Intelligent Public Address &
Voice Evacuation System



miniVES | midiVES

Compact plug-and-play
PA/VA system



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We make everyday life safer

Ambient System products are continually improved. All specifications are therefore subject to change without prior notice.

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