AMADEUS
ACTIVE ACOUSTICS

Functionalities & Technical Specifications

Preliminary notes

A complete active acoustics system consists of speakers, microphones, central technology (DSP, I/Os, etc.), software and services.
Amadeus Active Acoustics supplies the central processor and offers planning, installation and operational services for the entire system.

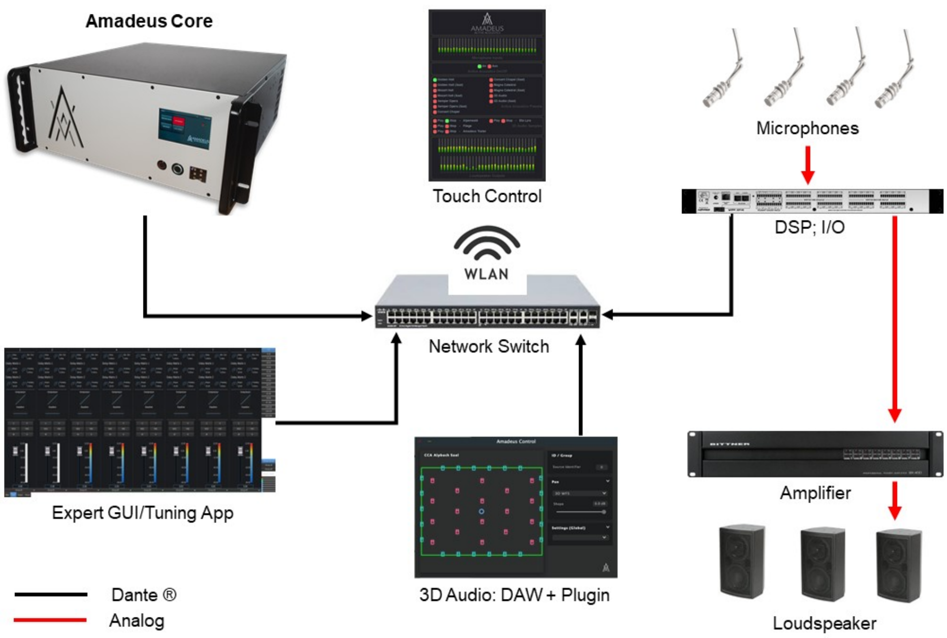

Figure 1: Diagram of a complete system

1. Functional Description

1. General description of the acoustic performance

Every space has its typical acoustical signature. This means that there is an existing reflection pattern and reverberation, which results from the temporal behaviour, the direction and the attenuation of sound reflections at the outer surfaces of the space, and their geometry. This signature must be taken into account and supplemented in the settings of the active acoustics to make the sound impression appear natural. An artificial, acoustically sounding room superimposed on the existing room acoustics should be avoided at all costs.

The digitally controlled electro-acoustic system consists of homogeneously distributed speakers and microphones, as well as a central processing and routing technology. A three-dimensional spatial model is stored in the processor for the calculations. This enables the enhancement of the natural reverberation times, the change in the reverberation level and the targeted generation of early reflections. The electroacoustic simulation and addition of natural room properties influences the absorption and reflection behaviour of the room surfaces (absorptive, reflective, scattering, sound-directing, etc.) and thus the entire sound propagation in the room, which is perceived naturally and, above all, without discolouration. Additionally, the system can help the musicians or performers hear each other better through early reflections. The system is consequently able to change the acoustics of different areas within the hall individually.

Extreme acoustic conditions can be used, for example, as an effect. The reverberation time can be extended continuously from very short (e.g. 0.1 seconds over the structural reverberation time) to very long (e.g. 3 seconds). All important acoustic parameters are freely and dynamically controllable and adaptable, for example, Reverberation time (T30), Early Decay Time (EDT), level of the reverberation, Strength (G), frequency weighting of the reverberation, time of arrival and direction of early reflections, Clarity (C50 / C80), Lateral fraction (LF), frequency weighting of early reflections, centre time (TS), etc. The electronic acoustic system does not lead to instabilities, noise, feedback or other disturbances in any situation. The natural spatial location of the sound source(s) is (are) not changed or disturbed by the system at any point.

2. Amadeus Core; Technical characteristics

The processing works as a hybrid combination of both "in-line" and "regenerative" systems. This means that both early reflections and reverberation are generated electronically (in-line technology), and the reflections and reverberations that occur naturally in the room, as well as the reflections and reverberations generated by the system itself, are picked up, processed and reproduced as "controlled feedback" (regenerative technology). Solely by combining both approaches is the entire, natural-sounding sound field generated. The processor offers configuration options to be able to influence different acoustic zones within a room at the same time. It works internally with a sample rate of 48 kHz. The Amadeus Core is equipped for a signal transfer of 128 I / O channels at 48 kHz via DANTE with a latency of 1 ms or less. The processor can process the signals from up to 128 inputs and generate individual signals for 128 outputs.

The processor is configured using computer software. The expert GUI consists of a uniform, graphical user interface which sets all parameters, such as controlling reverberation time, reverberation levels, position of early reflections, density and attenuation, 8-track parametric- and shelving equalizers, delays, routing, and microphone preamp settings. The software interface uses a client/server architecture to enable simultaneous access by several users. System settings are saved as ready-to-use presets on the device itself or via the web interface. The desired settings are saved on the system's permanent memory and can be loaded and displayed by the control software after the system has been switched off and on. The settings can only be accessed by authorized personnel.

3. Optional 3D Audio: “Amadeus Show Acoustics”

Optionally, the system can be expanded for 3D audio applications, and the installed speakers can be addressed via the existing Dante channels.

Sixty-four separate signal inputs allow 3D audio spatialization for live events or sound effects. Audio sources can be made audible in 3 dimensions anywhere in the room. Based on the stored 3D room model, live signals can be moved continuously, and audio files can be played. A plug-in can be easily integrated into standard DAWs to enable pre-production and mixing.

Up to 64 3D objects and a maximum of 128 outputs can be processed simultaneously. The 3D audio module works simultaneously and independently of the active acoustics. Optionally, delay or intensity-based panning methods for 3D audio processing can be addressed individually for each 3D object ("WFS-Light" or "Amadeus Pan"). The source size, i.e. the number of speakers used, can be varied. All speakers are always taken into account in the calculation. This means, for example, that with a focused setting (size = 1), only one loudspeaker is addressed from the desired direction. If the size is increased, more and more loudspeakers in the vicinity of the direction-determining system are addressed. Both the levels and the delays are calculated dynamically so that the location to the source remains and the overall level in the room does not increase. With a maximum setting of size =100, all loudspeakers are in use.

Loudspeakers can be grouped. "Virtual speakers" can be configured for standard surround sound reinforcement (e.g. 5.1; 9.1, etc.). Pre-produced 3D materials can be saved in the processor and displayed via the user GUI.

General control inputs, such as OSC, are used to connect external devices.

2. Technical Specifications

1. System processor (Amadeus Core)

The Amadeus Core contains the algorithms for calculating the early reflections based on the stored 3D model and the direction-related detection of the source positions. The processing is time-invariant without tone distortions or any noticeable latency. Also, a 128-channel reverb generator is included, which takes the sonic properties of the early reflections into account and can thus imitate the acoustic signature of the given room. All functions for routing the audio signals as well as the connection options for audio sources and sinks are included.

To achieve an optimal system level and to be able to carry out the necessary time corrections, the system is equipped with parametric equalizers and integrated signal delays. A high loop gain is achieved through decorrelation modules. Additional DSP functions include 8-track parametric EQs for inputs, outputs, reflection levels and reverberation, dynamic processing of the input signals and input/output mixer. The system is fully configured via a graphical user interface. The settings can be saved in the system's non-volatile memory with presets. The control and parameterization take place via Ethernet. The remote control takes place via the web interface.

The system inputs and outputs have digital formats for DANTE network connections. 128 channels of inputs and outputs are available.

The system includes redundant power supplies.
The software is based on operating systems and is delivered pre-installed on standard hardware platforms.

2. 3D Audio Extension “Amadeus Show Acoustics”

Extension of the system processor by 3D audio modules for playback or real-time positioning of sound sources.

- Delay or level based panning algorithms (WFS-light, Amadeus Pan), individually selectable for each 3D object
- Panning modes: 2D, 3D
- Apparent source size selectable for each sound source
- Web interface for preset control
- VST plug-in for standard DAW; Real-time positioning of virtual sources

- OSC protocol for control commands from external devices (e.g. tablet, mixer, etc.)
- 64 input channels / 16 output channels (optionally expandable to 128 outputs)

Summary (Specs)

General Specifications	
Terminals	DANTE In/Out; Remote Control (Ethernet RJ45); Service (Maintenance) RJ45; Option: MADI optical In/Out
Power supply	100-240Vac/ 8-4 A/ 700W/ 50-60Hz
Power supply (redundant)	100-240Vac/ 8-4 A/ 700W/ 50-60Hz
Installation	19" Rack mounting; 4RU; Box Dimensions 520mm(H) x 205mm(W) x 509mm (D)
Net Weight	approx. 15kg

Audio Specificationen	
Processing Channels (Dante or MADI)	128 IN/ 128 OUT
Example: Inputs x Outputs	IN1-64: 64 3D Audio Inputs(Show Acoustics Option), IN 65-128 64 Microphone Inputs, OUT 1-128 128 Speaker Outputs
Sample Rate	48kHz
Audioprocessing	I/O: 24bit, Signal Procesisng 32bit floating point
Synchronisation	DANTE input; Option: MADI

Internal Processing Specifications	
Input Stage	Phase decorrelation, Input Mixer, Dynamic processing, Spectral shaping, Input Grouping, Input Mute/Solo
Option: Show Acoustics	3D Audio Inputs; General Purpose Inputs
Acoustic Engine:	
Early Reflections	Time-structure and direction based on 3D Modell; Spatial, level, density and spectral shaping; grouping and group-processing for pre-defined areas of the room (e.g. stage, parterre, balcony…)
Late Reverberation	128 channel reverb; Decorrelation; reflection cluster (density, emphasis time, spectral shaping) Reverberationtime in frequency bands, density empasis time

Option: 3D Engine	64 3D Audio objects simultaneously on 128 speakers, 3D Audio mode switch (WFS light, Amadeus Pan) individually on each object simultaneously; source size; virtual speaker; bass management, gestural control, works simultaneously to the acoustics modul
Output Stage	Loudspeaker equalization, Output Mixer, speaker grouping, speaker mute/solo, speaker test oscillator
Control Interface	UPD; OSC; Midi;
GUI	Internal Webserver; Control, Scene (preset) storage and automation