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Design of a multichannel audio system based on A^{2}B architecture

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ABSTRACT
The main constraints when designing a multichannel audio system are management and routing of the great amount of audio data involved, and often also the cost. One of the principal issues with analogue systems, mostly employed in the past, was dealing with wirings. Nowadays, solutions are mainly digital and based on audio-over-IP solutions, which simplify audio routing. On the other hand, each device requires expensive electronics, such as FPGAs or processors, to manage these protocol stacks. Moreover, audio-over-IP protocols typically introduce latency (usually in the order of few milliseconds) which can limit the application field of the system. This paper presents the design of a multichannel audio system based on the Automotive Audio Bus (A^{2}B), and more specifically describes the core hardware unit which was developed for this purpose, named the A2B-DSP board. The proposed architecture guarantees a deterministic latency of 2 samples, and it makes the audio system modular, thus offering the possibility to realize different geometrical configurations. Number of channels ranges from 32 to few hundreds, making the system suitable for any room and spatial rendering technique, such as Wave Field Synthesis (WFS) or Ambisonics.

1. INTRODUCTION
Spatial audio techniques are gaining popularity thanks to the improvement of signal processing algorithms and availability of powerful electronic devices, which are more and more affordable. Spatial audio reproduction systems typically require a high number of loudspeakers to improve spatial accuracy [1],[2], usually at the price of higher electronics cost and bulky wirings. The employment of audio-over-IP technologies [3] simplify the cabling, but it requires expensive devices (typically FPGAs or processors) that contribute to increase the system cost, also from a development point of view.

This paper introduces an architecture for multichannel audio distribution systems based on the Automotive Audio Bus (A^{2}B) technology [4]. A^{2}B allows reducing the cost of the system, since the protocol is managed by dedicated low-cost transceivers that do not require software management, but only an initial configuration. In addition, A^{2}B guarantees a deterministic latency of just 2 samples (less than 50 µs at 48 kHz), as well as synchronization between devices.

2. CAPABILITIES OF A^{2}B SYSTEM
A^{2}B is a technology developed by Analog Devices for the automotive field. It is based on a digital bus capable of supporting up to 32 channels (32-bit wide) at 48 kHz, as well as power delivery (up to 2.7 W for each bus) and control over-distance (GPIO and I^{2}C commands). A^{2}B is a multi-node bus: a single bus can be composed by one main node and up to ten subordinate nodes. The maximum distance between two nodes is 15 m, while the total bus length is 40 m. Nodes communicate over an Unshielded Twisted Pair (UTP) cable, which is very cheap. Since the system will be used in the professional audio field, it was chosen to adopt XLR connectors and AES/EBU cables due to their large adoption.

A^{2}B allows reducing the system design effort since the access to the bus is completely managed by a dedicated transceivers developed by Analog Devices. The only required operation at the start-up of the system is a configuration of the transceivers on the bus, which can be performed by a simple and low-cost microcontroller. Even if redundancy is not supported natively by A^{2}B, the technology offers fault diagnostic, by which is possible to identify, localize, and isolate faults, while other nodes continue working. A^{2}B nodes are synchronized on the main node clock, and each node reconstructs its clock from the *superframe* (namely the data packet transmitted on the bus) transmission rate, that is the sampling frequency. Since each node reconstructs its clock from the main node clock, it is important that the clock source is jitter free [5].

For some applications, A^{2}B presents some limitations such as limited cable length and limited number of channels. Our system overcome these limitations by converting A^{2}B data to other common audio protocols (e.g., MADI, AVB, Dante, Ravenna, AES67 or USB UAC-2) [6]. A block diagram of the system architecture is shown in Figure 1.

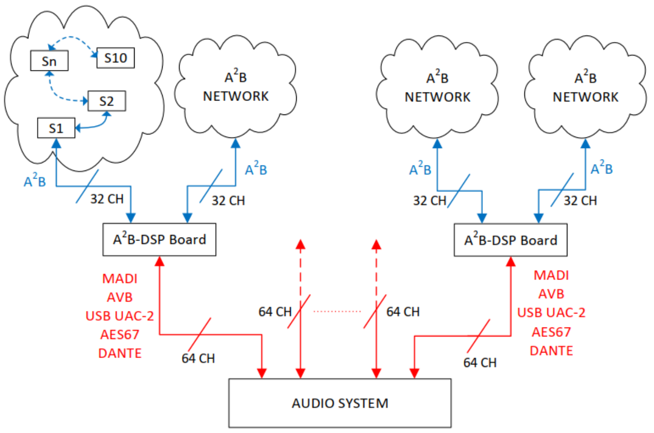

Figure 1 – Block diagram of the system architecture.

The system architecture is composed by three main parts:

- Audio system, such as a mixer or an audio interface.
- Conversion and processing board, namely the purpose-built A^{2}B-DSP in Figure 1.
- A^{2}B networks: each network is composed by subordinate nodes that communicate with different type of devices depending on the application. Some examples are amplifiers, processing units, pre-amplifiers, and microphone arrays.

The A^{2}B-DSP board has two main functions: protocol conversion and signal processing. The board provides the connection to two A^{2}B networks, for a total number of 64 signals, as well as other auxiliary input/output, as shown in Figure 2 and Figure 3.

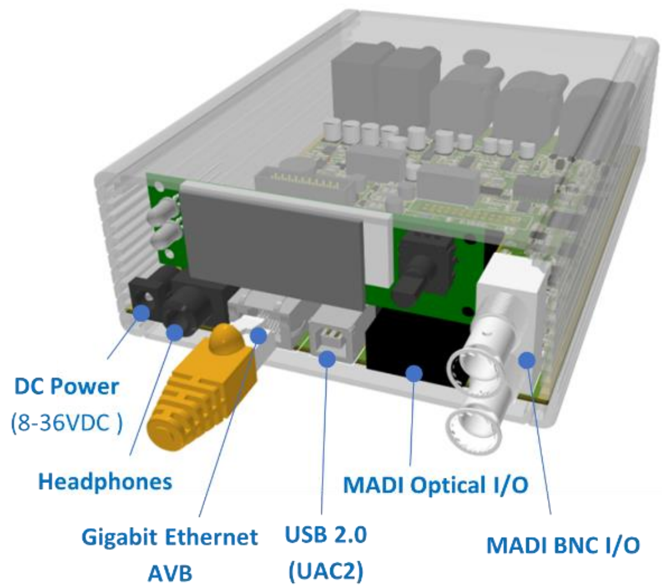

Figure 2 – A^{2}B-DSP connectivity, front view.

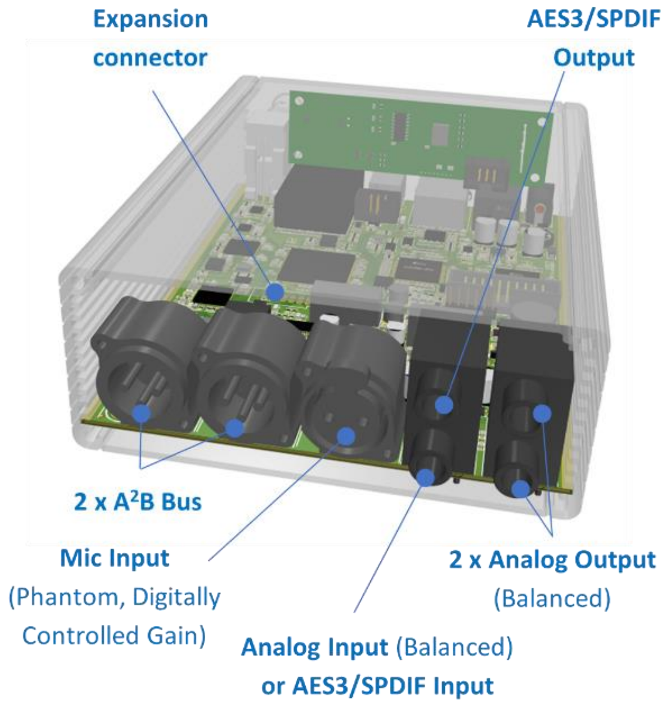

Figure 3 – A^{2}B-DSP connectivity, rear view.

3. FEATURES OF THE NEW A2B-DSP BOARD

This board, shown in Figure 2 and Figure 3, is the core of the proposed A2B system. It allows to drive the system through any general-purpose Audio over IP network, such as Dante, Ravenna, Madi, AVB, etc. by employing the corresponding firmware. An USB interface is also available (but limited to 32 channels) for quick setup and tests.

Apart the capability of interfacing the two A2B busses with the Audio-Over-IP interface, the A2B-DSP board provides many other useful features:

- Two Sigma-DSP processing units, which can be programmed for providing various types of processing.
- An FPGA which can be used for implementing different communication protocols
- Two analog inputs (one Mic, one Line) for performing acoustical measurements
- Two analog outputs, for driving analog loudspeakers
- Digital AES3/SPDIF input and output
- Madi Coaxial and Optical bidirectional interface
- A Giga-Ethernet socket to be used for Dante, Ravenna or AVB digital audio bidirectional connection.
- A robust input power socket, which can accept a wide range of voltages, and is virtually immune from electrical noise (so it can be used, for example, attached directly to the battery of a car)

The A2B-DSP board was designed from scratch and built in a small pre-series, for being used in several research projects involving the University of Parma, such as Sipario [6], and PHE [7]. A commercial version is planned for 2023 [8].

4. EXAMPLE OF A MULTICHANNEL AUDIO SYSTEM

The presented architecture is particularly suitable for realizing signal distribution in WFS listening rooms. This kind of system requires a huge number of loudspeakers that encircle the room. The number of channels involved is typically hundreds. Thanks to the modularity of the proposed architecture, it is possible to build systems of different sizes, which can be adapted to any listening room, from very small ones (e.g., 2×2 m, 64 channels, one seat) to very large one (e.g., 6×6 m, 192 channels, 40 seats). This is achieved just by increasing the number of A^2B-DSP boards, each of which provides 64 additional channels over two A^2B networks.

5. CONCLUSIONS

This paper has shown the benefits of the proposed architecture when employed for audio distribution in multichannel listening rooms. Research on spatial audio reproduction techniques gained popularity, but such systems are still expensive. The presented architecture aims to make more affordable the system if compared to other technologies. In addition, it offers a low, deterministic latency and the possibility to develop modular, expandable, and adaptable solutions.

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