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 data, cabling, and cost. This paper presents the design of an electronic board based on the Automotive Audio Bus (A^{2}B), aimed to realize multichannel audio digital systems. The proposed architecture guarantees a latency of 2 samples, and supports hundreds of channels, making it suitable for any spatial audio rendering technique, such as Wave Field Synthesis (WFS) or Ambisonics.

Keywords – A2B bus; Multichannel Audio Systems; Wave Field Synthesis

1. INTRODUCTION

Spatial audio techniques are gaining popularity thanks to the improvement of signal processing algorithms and availazbility 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, particularly in case of analogue systems. The employment of digital solutions and audio-over-IP technologies [3] simplify the cabling, but it requires expensive devices (such as FPGAs or processors) that contribute to increase the system cost, also from a development point of view. In addition, 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 introduces an architecture for multichannel audio distribution systems based on the Automotive Audio Bus (A^{2}B) [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 multi-node: 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), which is a low-cost cable. 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 may present limitations, such as a limited cable length and a 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], [7]. A block diagram of the system architecture is shown in Figure 1.

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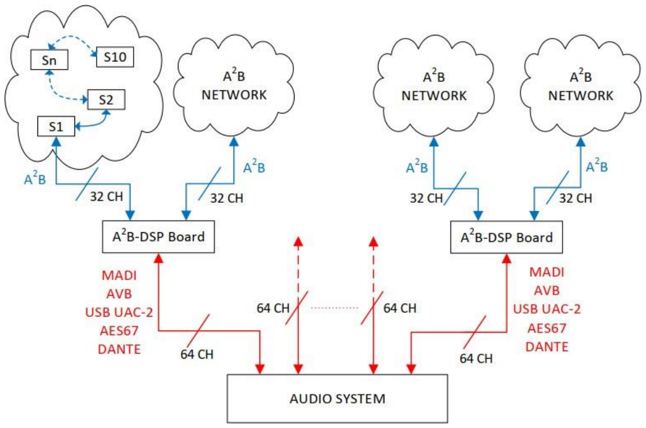

Figure 1 – Block diagram of the system architecture.

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^2B-DSP in Figure 1.
- A^2B 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^2B-DSP board has two main functions: protocol conversion and signal processing. The board provides the connection to two A^2B networks, for a total number of 64 signals, as well as other auxiliary input/output, as shown in Figure 2 and Figure 3.

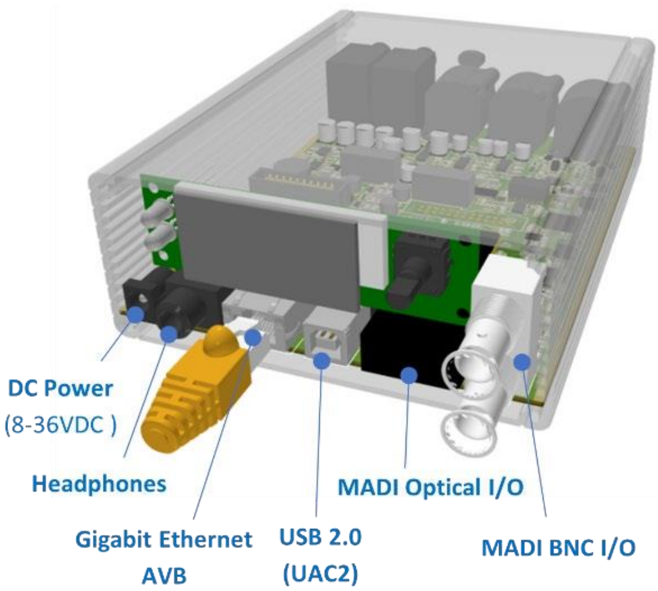

Figure 2 – A^2B-DSP connectivity, front view.

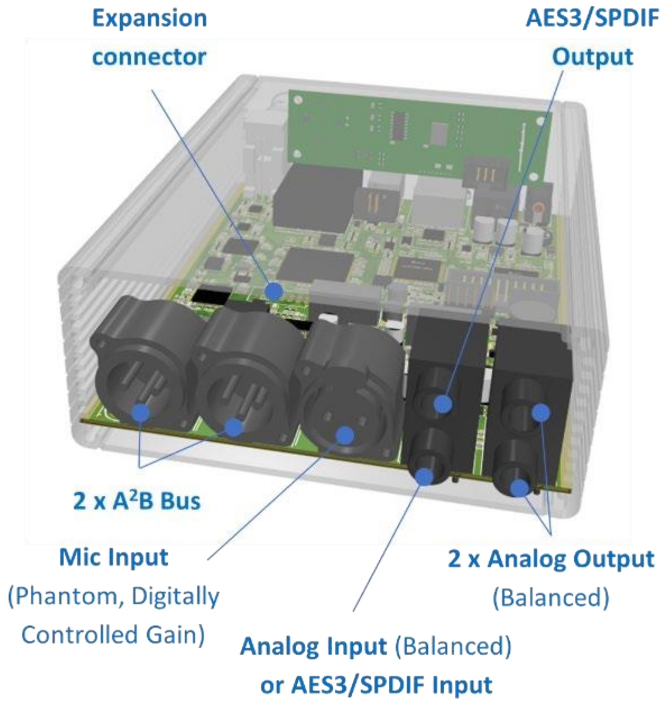

Figure 3 – A^2B-DSP connectivity, rear view.

3. FEATURES OF THE NEW A^2B-DSP BOARD

The board, shown in Figure 2 and Figure 3, is the core of the proposed A^2B system. It allows controlling the system through any general-purpose Audio over IP network, such as Dante, Ravenna, Madi, AVB. An USB interface is also available, even if limited to 32 channels. Additional features of the A^2B-DSP board are listed below:

- 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 analogue inputs (one Mic, one Line) for performing acoustical measurements.
- Two analogue outputs, for driving 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.

The A^2B-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 PHE – Past Has Ears [8]. A commercial version is planned for 2023 [9].

4. EXAMPLE OF A MULTICHANNEL AUDIO SYSTEM

The presented architecture is particularly suitable for realizing signal distribution in Wave Field Synthesis (WFS) listening rooms. This kind of system requires a huge number of loudspeakers, typically hundreds, that encircle the room. 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 them more affordable than other traditional technologies. In addition, it offers a low, deterministic latency and the possibility to develop modular, expandable, and adaptable solutions.

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