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A distributed Audio System for Automotive Applications

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ABSTRACT

With a trend to higher levels of drivetrain electrification and autonomous driving, the technology to increase audio performance is becoming a more significant factor of request. Instead of centralizing related signal processing in a single powerful hardware platform, distributing it in a more intelligent way can lead to several advantages such as optimized cabling, reduced weight, improved system scalability, performance and costs. The distributed audio system proposed in this work is connected to an automobile head unit that serves as human machine interface and media source. Portions of the data acquisition, signal processing and amplification are placed within distributed processing nodes. We present a realization with 34 loudspeakers and 16 microphones featuring seat-individual 3D audio rendition, in-car communication and further innovative use cases.

1 Introduction

Currently prevailing trends in the automotive market lead to requirements a prospective sound system has to cope with. At first, with increasing level of autonomous driving, more and more leisure time will be available for use cases raising the demand for high quality audio output and speech signal input – each individually and simultaneously on every seat. Additionally, the system has to deal with variable interior arrangements since seats may turn around or change their positions inside the vehicle cabin.

Secondly, electric vehicles' missing engine noise foregrounds road and wind noise in human perception – especially because electric vehicles tend to be more lightweight and therefore use less acoustic damping materials, which leads to an increased cabin noise level. Consequently, the demand for active noise cancellation evolves.

Both trends lead to overall higher system complexity with increasing number of speakers, microphones and other peripherals to be connected. Solving this task by utilizing a conventional system architecture based on one central electronic control unit would certainly lead to extremely complex and heavy wire harnesses and costly hardware platforms as well. Therefore, a new approach is needed comprising a revised architectural concept that allows connecting all components in a plain way at moderate costs.

Recent advances in audio and digital communication technology offer new opportunities: New low-latency automotive bus systems [1], [2] offer flexibly designable network topologies to transport audio and meta-data while class-D amplifiers offer high fidelity and efficiency as well as small footprint and digital connectivity [3], [4]. Additionally, embedded computing power is getting more inexpensive and available at various scaling levels.

The combination of these advances offers the convenience of solving the aforementioned problem by introducing a novel sound system architecture that is based on a distributed system approach.

The paper is outlined as follows: We give an overview of the distributed system, its scalable hardware building blocks and software processing modules in section 2. In section 3 we present a demo car implementation controlling 34 loudspeakers and

16 microphones and address associated use cases before we conclude with a summary.

2 Distributed System Architecture

2.1The Digital Bus

An example of a bus system suited for the needs of a distributed system is the Automotive Audio Bus (A^2B) [1]. It allows transmitting multiple audio channels downstream and upstream using a lightweight unshielded twisted pair connection with approximately 2 samples latency between the time division multiplex (TDM) inputs and outputs at the A^2B transceivers. The bus master provides the clock and the network slaves are daisy-chained nodes. In addition to its synchronous streaming capabilities, A^2B allows to tunnel asynchronous control data by offering two methods: An I^2C tunnel can setup the connected hardware and a 4-byte mailbox can be used to exchange control information with connected processing units (µCs or DSPs). Thus a fully controlled infrastructure can be provided and interface software wrappers in the style of AES-70 [6] become possible.

2.2 Scalable Hardware Modules and System Architectures

Hardware modules are assembled out of a Base Module (BM) with a bus interface, class-D amplifiers and A/D converters for sensor inputs. BMs may be extended by daughter boards:

- A daughterboard with a small-scale fixed-point DSP, denoted by Audio Module (AM).
- A daughterboard offering a high-performance multi-core floating point DSP and additional bus interfaces to allow for scalable connectivity and high computing power (denoted by Computing Module, CM).

Due to their small dimensions, the HW modules can easily be packaged near associated peripherals, such as loudspeakers or microphones, which results in significantly reduced wiring efforts and improved signal quality. Because of inherently short speaker wires low gauges may be chosen with negligible weight and costs. Then the connected amplifier is enabled to deliver its power to the speaker via high currents instead of high voltages which eliminates the need for expensive heat and EMC problematic DC/DC converters and allows operating the amplifier directly at the vehicle's power supply.

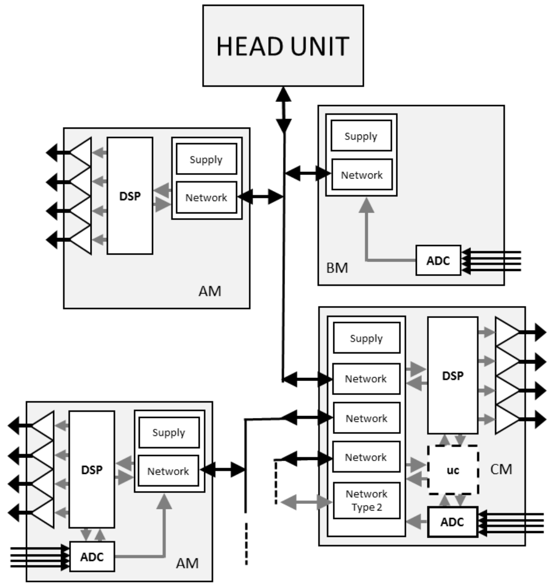

Figure 1. Distributed system with multiple busses and adapted nodes. The head unit (HU) is connected to a Computing Module (CM), which includes a multicore DSP and controller. It receives audio from the HU and the ADCs, executes the main processing tasks like entertainment and ICC and deploys the processed audio streams.

2.3 Processing Modules

2.3.1 *Entertainment*

In its current realization, entertainment processing features a combination of audio channel upmix [7], equalization modules, a bass management stage and a limiter.

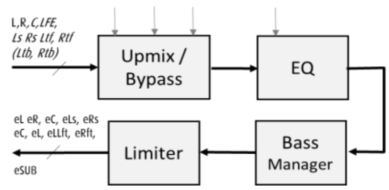

Figure 2. Entertainment processing blocks.

Entertainment processing is executed in the HU or in a CM node of the distributed audio system. The number of audio channels can vary from 2 (no upmix), 3 channel stereo (only center channel upmix), 5.1 channel surround and 5.1+2 or 5.1+4 channels for 3D audio. Equalizing pre-sets are tuned to flatten or boost the Car-Hifi-Curve to optimize playback for different driving scenarios. The bass manager's task is to redirect low frequency components of all channels to a subwoofer channel. The limiter prevents clipping of audio signals.

2.3.2 *In-Car Communication*

In-car communication (ICC) captures a passenger's voice, suppresses feedback, ambient and entertainment signals and transmits the clean speech signal to another passenger's loudspeaker with only low processing delay. This enables easy communication at high ambient noise levels encountered at high speeds and open roof convertible driving scenarios. ICC building blocks include sampling rate conversion, best microphone selection out of multiples per seat, speech detection, echo cancellation and noise suppression. Thereafter a routing and mixing stage ensures that the signal is directed to the desired listener depending on the ICC modes, such as driver↔ passenger, front↔rear and telecommunication links. The processing delay is kept sufficiently low to ensure naturally comfortable and stress-free communication. Details are given in [8] and [9].

2.3.3 *Loudspeaker Management*

An exemplary programmable model is shown in Fig. 3. Eight input channels can be processed to four output channels. There are two programmable channel routers and mixers. A first mixer is used to combine low delay speech and the entertainment signals. The mixer at the very end can be activated for mixing in very low latency signals like band limited anti-noise signals. Gain and delays are used to compensate for different distances of the loudspeakers and can be updated in real-time depending for the selected listening zones. A filter stage of up to ten cascaded biquads IIR filters is used to adapt to the loudspeaker transmission characteristics. The minimum processing delay is two samples. Usually loudspeaker management processing is performed within a small scale DSP.

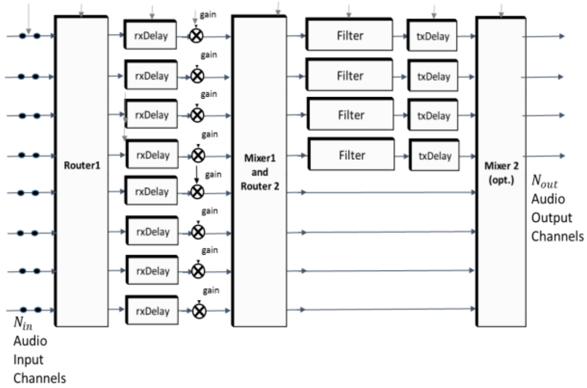

Figure 3. Programmable processing model of the loudspeaker management and signal mixing stages

2.3.4 *Loudspeaker Linearization*

Woofer linearization reduces the total dynamic distortion of a loudspeaker at larger diaphragm displacements and extends its usable displacement range. The other way around, smaller designs become possible. Additionally, strong variations of temperature (very common in cars) change the characteristics of the loudspeakers' mechanical suspensions. This also influences the speaker´s transfer function. By monitoring the loudspeaker current, adaptive regularization of the transfer characteristic becomes possible to ensure application-specific transfer characteristics as well as secure operation in all considered circumstances. This regularization processing can easily be implemented within the DSP of a network node. The distributed architecture allows direct neighbourhood to the speaker and this ensures optimal performance.

2.3.5 *Road Noise Cancellation*

Acceleration sensors and microphones placed around the vehicle body can be utilized to accurately determine the relevant road noise signal portions. For applying the anti-noise signals, subwoofer, the door's speakers and loudspeakers placed close to the passengers' ears can be utilized. The sensor data can be transmitted via the bus network and processed using an algorithm similar to the filtered-x least mean square (FxLMS) [10] to generate the anti-noise signal. Special fast ADCs and low-latency signal transfer paths can be used to keep the signal transfer delay small enough to ensure significant noise reduction up to the kHz region.

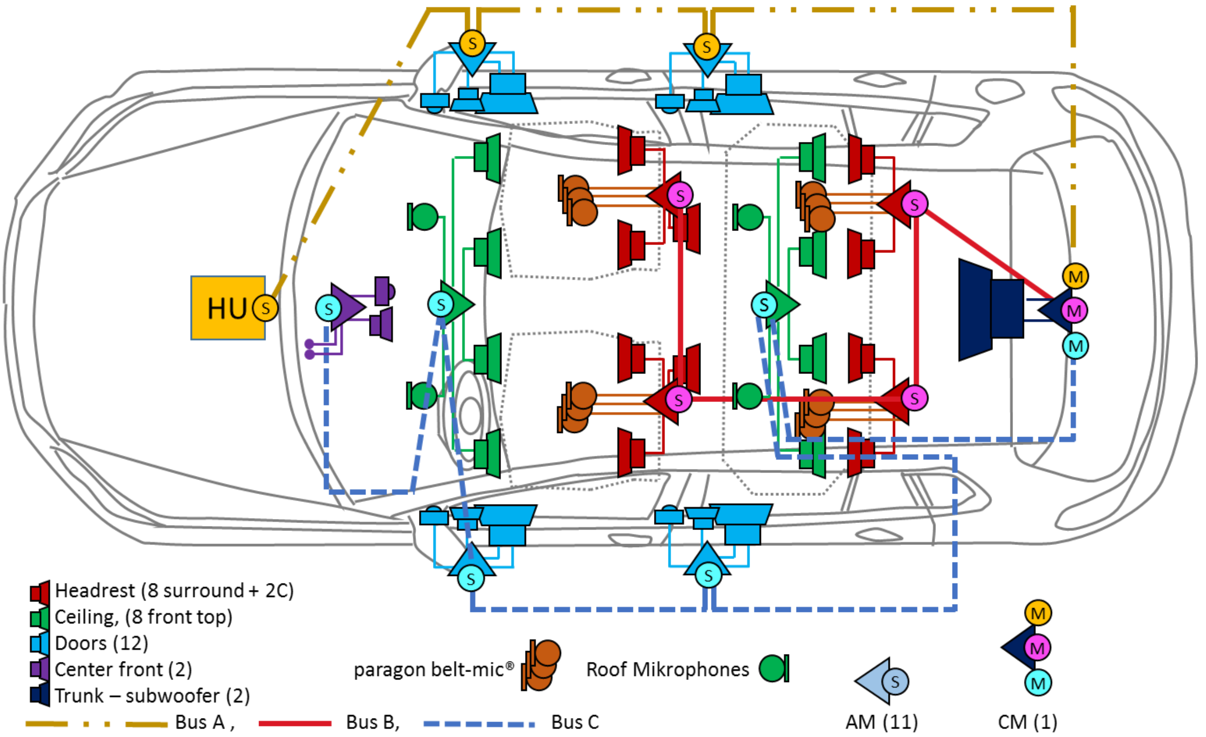

Figure 4. The demo car is equipped with a distributed system linked by 3 busses (A, B, C). The HU serves as a HMI, media source and telecommunication link. Entertainment and ICC processing is executed in a CM node. AM nodes close to loudspeakers carry out the loudspeaker management and mixing of speech signals. Class-D amplifiers drive 34 loudspeakers. 16 microphones are available for ICC and hands free telecommunication.

3 A System for Demonstration

3.1 System Description

The structure of the demo car setup is shown in Fig. 4. A woofer, midrange and tweeter solution is placed in each of the door panels, a two-way center channel solution mounted within the top of the dashboard and a double subwoofer is hidden within the spare tire cabinet of the trunk. Loudspeakers to support individual sound zones are used for the two surround and top-front channels: There are two speakers mounted within the roof in front of each seat. Two speakers are placed in each headrest, inside the foam body and hidden beneath perforated leather, utilizing high-performance speaker equalization to overcome the influence of the packaging situation. Additional center channel support facing towards the back row is placed in the driver and co-driver headrest. The speaker setup supports 5.1+2 top front multichannel audio formats directly. 5.1+4 audio needs to be processed: Each loudspeaker is connected to a class-D amplifier as an element of the processing nodes, 11 AMs and 1 CM. There are four top microphones placed in the roof jacket in front of each seat: The safety belt of each seat is equipped with three

microphones (paragon belt-mic®). These microphones are connected via AD converters of the system´s processing nodes to the digital bus.
The CM is equipped with a dual-core DSP plus a microcontroller processing unit. An ARM core is used to handle the interrupt-driven mailbox control flow initialized by the HU. While one of the DSP cores is used for the entertainment signal processing (see section 2.3.1), the other one is used to perform additional features like in-car communication (see section 2.3.2) with a much shorter processing delay.
Processed entertainment music and ICC voice signals are sent to the remote nodes (AMs) close to the loudspeakers in synchronous streams. The receiving nodes are equipped with a fixed point DSP to handle the mailbox command processing and the loudspeaker management processing; including mixing of the ICC signals (see section 2.3.3).

	Distributed System	Centralized System
Length cabling Microphones	9.8m	41m
Length cabling Loudspeakers	15.3m	117.5m
Length cabling A^2B	24.5m	
Length cabling Power supply	24.5m	
Length cabling Overall	97.6m	158.5m
Weight Overall	15.6kg	21kg

Table 1. A comparison of cable harness and related weight of the distributed system and a conventional centralized design. It shows a weight reduction of 26%. Both systems with 34 loudspeakers, 16 microphones and up to 2000 W amplification power. The weights of the central booster amplifier is incorporated with 4.4 kg (including a DC/DC converter), the weight of a single AM node with 0.3 kg and a CM with 0.42 kg with enclosures. Cable diameters are aligned to tasks and signal power.

3.2 Use Cases

3.2.1 *Seat Individual 3D Audio*

The top and surround channel loudspeakers are placed close to each passenger which builds up the 3D image. The frontal image is stabilized and the stage is slightly elevated to match the windscreen. The close surround channel loudspeakers allow a direct path to the listener's ears and special processing allows for virtual height from behind. In addition, different loudness zones can be selected to adjust to the listener's preferences.

3.2.2 *In-Car Communication and Hands-free Telecommunication*

Microphones integrated into the seat belts (paragon belt-mic®) capture the passenger's speech with high SNR to be processed by ICC. ICC modes like driver↔passenger, front↔rear and support for all seats allow easy communication and low disturbance. High privacy and intelligibility is given when using the headrest speakers for playback and the most natural experience is gained when using the top loudspeakers, especially for the back row. Hands-free telecommunication is enhanced in the same way. The same ICC technology enables a private phone call using belt microphones and headrest speakers or a multi-person conference routing the caller to attendees' loudspeakers and their mixed speech to the HU link.

3.2.3 *Road Noise Cancellation*

The usable frequency range of active noise cancellation can be extended to the kHz range using headrest speakers. Road noise cancellation processing (see section 2.3.5) can be performed in a CM.

4 Conclusions

A distributed hardware and software architecture offers flexibility in terms of adaptation to different upcoming tasks and configurations while the automobile's head unit operates as a human machine interface and media source. The proposed audio system can be realized from a simple entry level system to an ultimately enhanced high-end network, comprising only a few to a very high number of loudspeakers, microphones and sensors respectively by simply combining different numbers of only a few variants of basically identical hardware building blocks. Validation and qualification costs can be reduced due to the decreased number of variants.

Wiring efforts and weights can be minimized, while electric efficiency and EMC can be enhanced due to the omission of DC/DC converters.
Functional elements like multichannel entertainment, loudspeaker management and linearization, in-car communication and road noise cancellation can be computed by distributing the computing task over a plurality of processors. An additional benefit besides the flexible system scalability is a reduction in wire harness complexity compared to that of a centralized system with same performance. This is demonstrated in a demo car as a realization example, wherein we implemented a distributed audio system comprising 12 HW modules, 34 loudspeakers, 16 microphones and other peripherals. This system currently features seat-individual immersive 3D sound, seat-individual hands-free telephony and ICC at a performance level ready for series production. In the near future loudspeaker linearization and road noise cancellation will be approved for vehicle demonstration. While the former will be utilized to reduce overall loudspeaker HW and packaging costs, the latter will be utilized to further enhance the performance of the seat-individual audio setup in order to extend the volume varying single-content listening zones to multi-content listening zones.

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