Audio Network Based Massive Multichannel Loudspeaker System for Flexible Use in Spatial Audio Research

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ABSTRACT

One of the main research areas in the field of spatial audio is sound field synthesis. These techniques use arrays of loudspeakers to (re)create the sound field of a virtual source in a desired listening area. They are often termed as massive multichannel systems, because of the number of loudspeakers in practical applications is quite large. Such systems require an individual computed signal for each loudspeaker. In this paper we present a massive multichannel system consisting of 640 loudspeakers. The system is based on an audio network distributing the audio signals to a grid of amplifiers. It will be shown that the use of the audio network allows control of error-free operation for the entire system. The paper starts with a discussion about the concept and design-related decisions for such a system. After that, the realization will be described in detail. The features of the system are illustrated with an example application.

1. INTRODUCTION

Since the CARROUSO project in 2001, the development of spatial sound reproduction techniques is one of the main research areas at Fraunhofer IDMT [3], with a focus on techniques for synthesis of sound fields and virtual sources by using loudspeaker arrays. The most common ones are wave field synthesis and (higher order) ambisonics [2, 5, 7]. Practical implementations of these technologies are often termed massive multichannel systems due to their requirement of up to hundreds of loudspeakers.

In order to develop new algorithms, a reliable massive multichannel loudspeaker system is needed. Due to the sheer number of loudspeakers involved, the maintenance and configuration of such a system should be as easy as possible. With a particular focus on a dedicated setup that is used for current research activities, we present a prototype system that provides all these features by the use of an audio network. It consists of 504 loudspeakers stacked as a wall, expanded by a circular speaker array surrounding the auditorium, adding further 100 loudspeakers. The presented massive multichannel system is flexible, easy to control and scales up to 640 loudspeakers.

Basic requirements for such a system will be defined in section 2. Section 3 describes the concept of the audio distribution within the system. Afterwards, the global system architecture is explained. In sections 5 and 6 the practical realization is described. The developed rendering environment, which provides the audio signals will be described in section 7. An example application using the massive multichannel system is given in section 8. The paper ends with a short conclusion of the presented work.

2. REQUIREMENTS

Each spatial audio reproduction technique requires a specific loudspeaker setup. Therefore the loudspeaker system needs to be modular and adaptable to several arrangements including two-dimensional setups (linear, rectangular and circular) as well as three-dimensional ones (planar and spherical). It should be possible to use all 640 channels of the system simultaneously, as well as to split the system into multiple independent units.

It is common practice for the creation of loudspeaker signals to use consumer PCs upgraded with professional, digital sound cards supporting the MADI (Multichannel Audio Digital Interface) protocol [1]. Consumer PCs are powerful, easy to customize and provide a good cost-performance ratio. The audio input format for the system should be the MADI protocol.

In the context of spatial sound field synthesis, phase coherence among the speakers is vital, because it directly influences the positioning accuracy of the virtual source. The latency variation among the speakers is required to be less than one period at a sample rate of 48 kHz.

During the process of developing sound algorithms, incomplete mathematical methods and software bugs might lead to uncontrolled and unexpectedly high loudspeaker signals. Because many loudspeakers are active at the same time, these signals might be strong enough to damage the hearing of the development engineer. Therefore the gain of all speakers is required to be adjustable with one central control that acts independently from the connected audio input system.

Furthermore, it should be possible to adjust the characteristics of each loudspeaker individually. One reason for this is the need of equalization. Depending on the fitting of a loudspeaker, e.g. in a linear array, its transfer function needs to be adapted.

Despite the large number of loudspeakers, the system has to be reliable and its handling needs to be as easy as possible. This requires user transparent wiring, short cable run and a simple way to determine the correct functioning of the system.

Last but not least the costs of the entire system are also a significant point.

3. AUDIO DISTRIBUTION CONCEPT

A typical audio reproduction system consists of a source, an amplifier and a loudspeaker, interconnected by cables. This simple scheme applies to a single channel as well as to the system described in this paper. The large number of 640 loudspeakers, used in the presented system, makes a huge difference and demands a new concept.

First and foremost, the number directly affects costs and complexity of the system. The brute force design would be to repeat a single channel design to the number of required channels. This approach ignores potential opportunities for cost savings, structuring and simplification. Therefore a different transmission chain is needed.

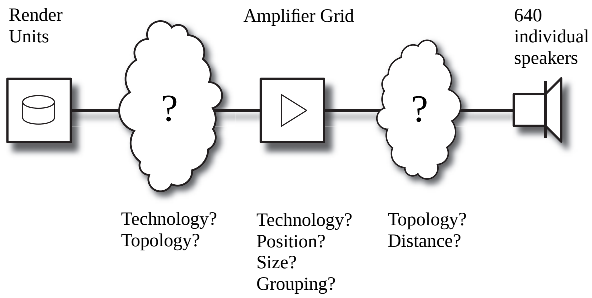


Figure 1: The tasks to be considered in the design of a massive multichannel loudspeaker system.

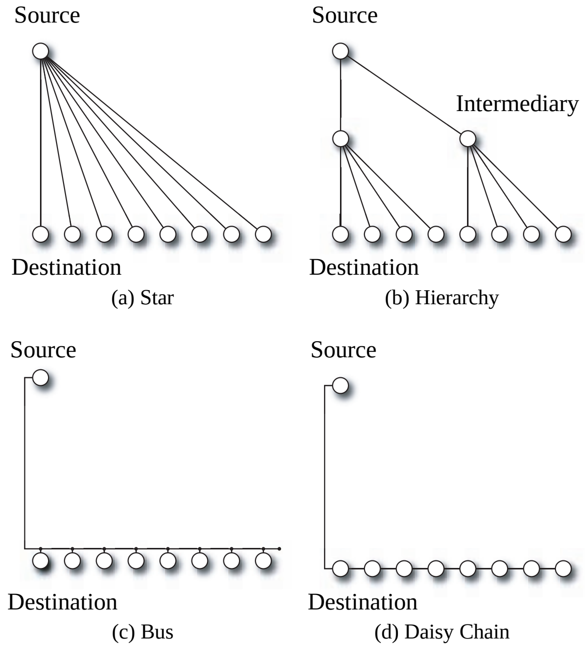
A schematic diagram of the introduced design challenges is shown in figure 1. The use of an audio network offers a suitable solution for these tasks. Its architecture plays a major role in the distribution of provided audio from render units to the speakers. Especially technology and topology of the network influences important requirements such as cable lengths, costs and flexibility. Other prominent points are the size, physical position in the network and grouping of the amplifiers.

It is natural to divide the system into segments, each assigned to 64 channels, because the audio signals generated by the render units are delivered using the MADI protocol. In such a segment the incoming signals will be distributed to a grid of amplifiers by a network to be defined and characterized in the following.

3.1. Topology

Seen from a systematic point of view, there are different choices for distributing audio signals to the amplifiers. These are depicted in figure 2. It is required to use as little cable as possible and to provide reliable operation. In this respect the star topology shown in figure 2(a) as well as the hierarchical topology shown figure 2(b) are both disadvantageous: total cable length, complexity and costs exceed that of all other solutions.

The bus topology shown in figure 2(c) consumes a minimum of cable run, because it uses a single common wire to connect all devices. It inherently supports the transmission from a single source to multiple, distributed destinations. A difficult problem to overcome is the realization of the bus taps. In this regard, we insisted to avoid inferior T-type connectors. Another disadvantage is the lack of implicit channel assignment. The bus wire carries information for all audio channels and each device has to



(a) Star (b) Hierarchy (c) Bus (d) Daisy Chain

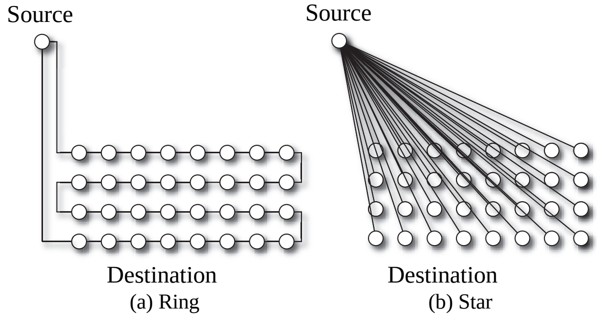
Figure 2: Common types of device connections for signal distribution from a single source to multiple destinations.

pick up the correct channel(s) on its own. Bus solutions are incapable of providing an elegant and error-proof solution in this regard.

In a daisy chain topology, illustrated in figure 2(d), connections are end-to-end. Each device is connected to the next, it receives the information for all channels and, while forwarding and refreshing it, picks up the relevant channel(s) for local deployment. Because each device has a fixed position in the chain, channel assignment is implicit and no administration is needed.

If a single connection in a daisy chain fails, the rest of the chain is also affected. This is of course an unacceptable behavior. The problem can be overcome by adding a redundant connection from the last device in the chain to the first one, as illustrated in figure 3(a). The daisy chain has been modified to form a ring.

On failure of one connection the ring reverts to a still functional daisy-chain, provided that the links of the ring are of bidirectional (full-duplex) nature. Additionally, full-duplex links allow local communication between neighbors e. g. bilateral monitoring of link integrity or performing media latency measurements. Figure 3 shows the star topology in comparison to the ring concept. It is obvious, that a ring of daisy chained devices is a very simple setup anybody can overview, even at a huge device count, while a star topology might become very confusing, especially in a two-dimensional array.



(a) Ring (b) Star

Figure 3: Direct comparison reveals that star topology (b), although commonly used in audio applications, leads to clumsy wiring, if interconnected devices are arranged as an array. A ring (a) simplifies interconnection. Redundancy ensures all nodes are still working, should a single connection in the ring fail.

We have chosen the ring topology as it best meets the given requirements.

3.2. Link Technology

It would have been natural to choose MADI as link technology, because audio is delivered in this format anyway. There are two standardized types of connection for transmission of MADI: coaxial cables with BNC-connectors and optical fibre cables using SC-connectors.

Table 1 compares the costs of possible interfaces and media in detail. The values represent averaged prices taken from typical offerings of international distributors namely Avnet, Digikey and Farnell. The table reveals that ethernet links are available at only a fraction of costs compared to MADI. Especially the costs for the optical fibre socket disqualifies an optical MADI solution. Media costs make the crucial difference.

In a competitive ranking that includes handling and reliability as well as availability, ethernet technology finally must be favored: Ethernet patch cables are ubiquitous, cheap, available prefabbed in any length and quality.

As a consequence, ethernet links were chosen as the preferred technology for distributing audio to the amplifier grid.

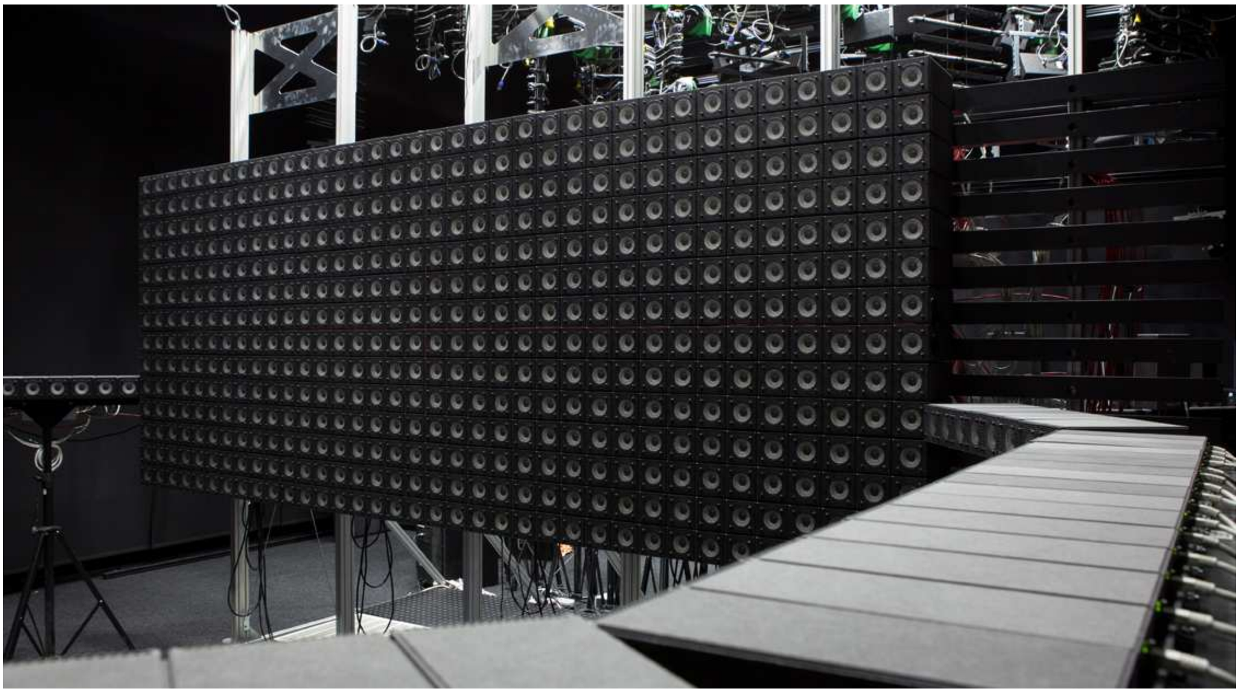


Figure 4: Image of the multichannel loudspeaker system, installed in the special acoustical laboratory at Fraunhofer. The system consists of 504 loudspeakers stacked as a wall and a surrounding circular array consisting of 100 loudspeakers.

4. GLOBAL SYSTEM ARCHITECTURE

This section provides an overall view for the practical realization of the massive multichannel loudspeaker system. Figure 4 shows this installation in a special acoustic laboratory at Fraunhofer IDMT.

4.1. Audio Transmission Chain

In figure 5 a schematic diagram of the audio transmission chain is shown. Audio signals from any external device are forwarded to the render units using MADI streams, where each render unit gets an identical input signal. Depending on the application, the render units run a specifically configured software (defined in a data flow graph). The generated signals of all render units are transmitted to the network control unit (NCU). The NCU converts the MADI signals to a network-specific format and supplies 10 rings of network amplifiers.

4.2. Render Units

For the generation of the individual loudspeaker signals, consumer PCs with quad-core processors are used. Each PC is equipped with two professional digital sound cards providing 128 audio output channels (individual loudspeaker signals) on two optical fibre lines (MADI) at a sample rate of 48 kHz. These are connected to the network control unit (NCU). An additional PC is used to configure and control the render units via ethernet.

4.3. Network Control Unit

The network control unit provides the link between the render units and the audio network by offering ten fibre MADI inputs and ten ring outputs. MADI audio information is transferred to the network links in a bit-transparent manner. From here on audio is combined with a communication layer that allows observation and control of all network units from this central unit. The NCU also provides digital synchronization for all audio channels, render units as well as network and digital amplifiers. The power stages of the amplifiers can be started and stopped using a single button. An attenuator on the front side of the NCU serves as a master volume control for all speakers. Additionally, a panic button is available to disable the system immediately in case of emergency. Like all the other components involved, the NCU does not require a fan.

Interface	ADAT TOSLink	MADI SC	MADI BNC	Network RJ45
Socket	2.50 €	23.00 €	0.25 €	0.25 €
Isolator				0.75 €
Tranceiver		2.00 €	4.00 €	2.00 €
Total	2.50 €	25.00 €	4.25 €	3.00 €
Channels	8	64	64	128
Costs [€/ch]	0.32 €	0.40 €	0.07 €	0.03 €

(a) Physical layer link costs

Medium	ADAT TOSLink	MADI Optical Fibre	MADI Coaxial	Network CAT5
Plug	1.50 €	2.00 €	0.75 €	0.10 €
Costs [€/m]	1.40 €	2.90 €	0.80 €	0.50 €
Costs including two plugs [€/10m/ch]	2.25 €	0.38 €	0.14 €	0.01 €

(b) Medium costs

Table 1: Physical link and media costs comparison between ADAT (Alesis Digital Audio Tape), MADI and Ethernet.

4.4. Network

One ring of daisy-chained digital amplifiers is capable of transmitting 64 audio channels. It starts and ends at the NCU and thus provides redundancy. For controlling the state of the audio network independently from the computer, the NCU offers a small LCD display. With its help, the status of wiring and redundancy can be monitored. Detection of cabling errors is indicated in a very intuitive way. This allows to control even a large number of loudspeakers.

The network also provides a communication layer, which is used locally and globally. Global tasks are controlled by the NCU, like setting the global volume or starting and stopping the amplifiers. It is also possible to address each individual network amplifier in order to read, modify and write an extended set of DSP controls. This includes sets of biquad filters, dynamics and volume, but also parameters for operational control.

Network nodes also communicate locally. This is provided to measure and compensate link latency, to analyze link integrity, to determine the own address in the daisy chain and to guarantee synchronous operation.

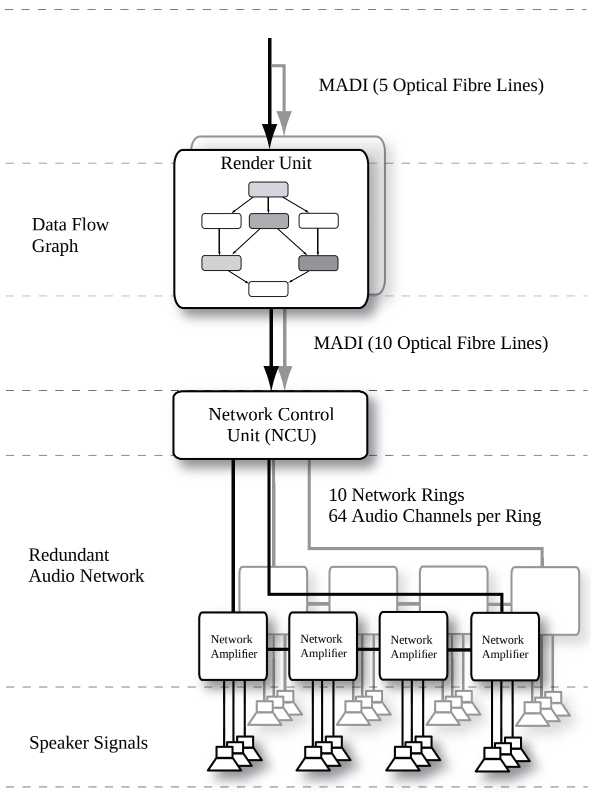


Figure 5: Schematic diagram of the audio transmission chain. Five render units provide the loudspeaker signals which enter the NCU on MADI streams. The NCU converts the MADI signals to the audio network protocol, adds a communication layer and distributes them to 10 individual daisy chains. Each daisy chain is connected to four network amplifiers and transmits 64 audio channels with full redundancy. Every network amplifier takes 16 assigned audio channels and transmits them to the loudspeakers.

Latency through the entire network is below 4 sampling periods. The maximum deviation of locally synthesized audio clocking along the network nodes is less then 10 ns.

5. NETWORK AMPLIFIER

The NCU transfers the digital audio signals to a grid of distributed amplifiers. Each daisy chain ring is connected to four network amplifiers. Leaving the digital domain for the purpose of amplification is disadvantageous, requires additional digital to analog converters and should be avoided.

5.1. Pure digital signal path

In a direct feature comparison focussing on size, costs and weight, analog amplifier concepts show no advantage at all against solutions based on Class-D concepts. The power range in this given application is defined to be 10 - 20 W per loudspeaker channel. This allows to choose from chipsets designed for consumer applications, where a broad range of superior solutions exists. All offer significantly reduced size and efficiency at a fraction of cost compared to any analog concept. These chips provide an excellent sound quality.

5.2. Integration

Driven by market demands for home cinema audio systems, many solutions offer integrated chips with six or eight audio channels. Additional features include filtering, dynamics and level control, making external DSP chips and the associated development efforts and costs obsolete. A Class-D concept of this kind was implemented, because the ability for filtering and level control is one basic requirement to adjust each loudspeaker channel individually. In the end, all DSP-related features were realized at no addditional expense. Furthermore, the use of a FPGA accommodates many functions. It is responsible for all duties regarding the network (MAC functionality), sampling frequency synthesis and audio decoding. An integrated microprocessor core serves for all communication tasks.

5.3. Resource sharing

By designing a printed circuit board (PCB) that feeds eight channels, cost per channel is reduced, because interface and controller chips are shared for these channels. In the presented concept, two PCBs are combined into one fan-less housing with an appropriate power supply to further reduce cost. Figure 6 shows the resulting 16-channel network amplifier.

5.4. Positioning

The physical position of the amplifiers in the signal chain is of great importance. They should be positioned very close to the loudspeaker wall in order to keep speaker cables short. This approach also simplifies wiring and reduces the overall weight.

Figure 7 shows the network amplifiers mounted at the back of the loudspeaker wall. Eliminating speaker cables is not an option, as it would interfer flexibility of speaker setups. For example, using active speakers and thus virtually avoiding speaker cables, requires an extra power connection per speaker. The developed network

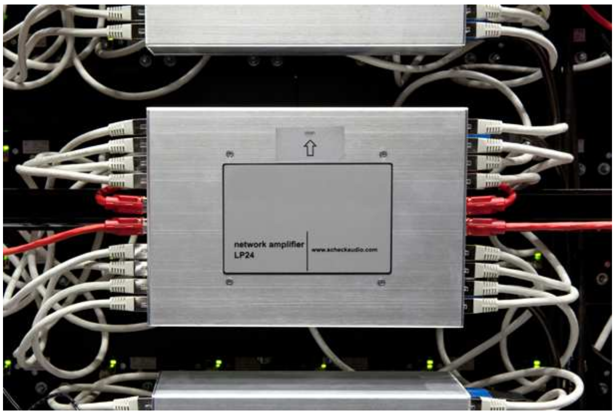


Figure 6: Close-up of the 16-channel network amplifier module. Grey patch cables connect the amplifier with the loudspeakers. The red patch cables are the links for the daisy chain. A green LED on the rear side of each loudspeaker indicates a proper connection.

amplifier module is small enough to be mounted directly behind the loudspeakers as depicted in figure 7 without impairing access to their connectors. This enables the use of very short speaker cables.

5.5. Speaker Cabling

Cables of various length must be interchangeable and must allow quick adjustment of the speaker arrangement. Reliable connections must be ensured, broken connections must be signaled. These requirements cannot be fulfilled with a usual two-wire speaker cable. Encouraged by the benefits of standard CAT5 cables its use as a speaker cable was evaluated (see table 1). One twisted pair of such a cable is reserved for communication with a management circuit, leaving the remaining three pairs for parallel connection between amplifier and speaker. Because an average power of only 10 W is specified per loudspeaker channel, this uncommon use of CAT5 cables is more than justified. The specifications in the Power over Ethernet standard support this decision [8]. The loudspeaker cabling using standard CAT5 patch cables can be seen in figure 6 and 7 (grey cables).

6. SPEAKERS

One fundamental component of such a massive multichannel loudspeaker system is, of course, the loudspeaker itself. In the context of spatial audio research, the interspace of the loudspeakers defines a so-called spatial aliasing frequency, which is an upper limit to control

Figure 7: Back view of the multichannel loudspeaker system. The network amplifiers (grey boxes) are mounted directly behind the loudspeakers.

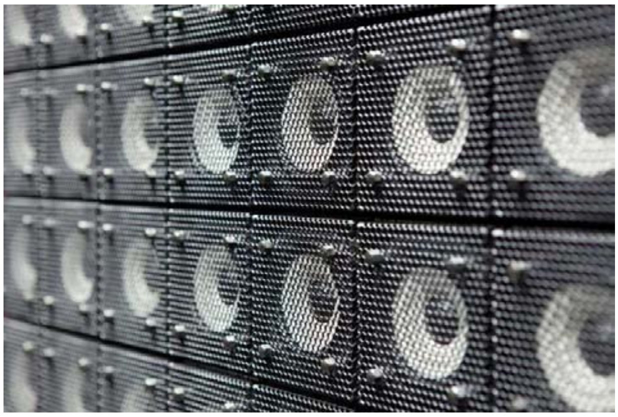


Figure 8: Close-up of individual speakers, arranged as a speaker wall. The interspace between the centers of two loudspeakers is 0.1 m .

the wave field [4]. Spectral components of loudspeaker signals above the aliasing frequency will result in non-coherent wave fronts of the virtual source. For this reason the restriction of the loudspeakers diameter is a significant decision with respect to overall performance.

Driven by the requirement for an aliasing frequency that is as high as possible, the width and height of the loudspeaker was defined to be 0.1 m. In [9] a loudspeaker panel is described that leads to a higher spatial aliasing frequency. However, the use of such a panel is not the best solution in our context, because here a small loudspeaker interspacing is needed in both, the horizontal and vertical direction.

A small enclosure only allows the fitting of a small loudspeaker driver unit. The smaller the cone of the loudspeaker driver unit, the more ineffective is its radiation at low frequencies [10]. Therefore, the dimension of the cabinet of 0.1 m is a compromise, that allows a fitting of a 3.5" loudspeaker drive unit. We used the VIFA TG9FD-10-08, which is a high quality full-range loudspeaker [12]. Due to the high number of 640 loudspeakers, the weight of the loudspeaker should be as low as possible. For this reason the enclosure was factored using aluminum profiles having a wall thickness of 3 mm. All loudspeakers are surrounded by a damping material to attenuate resonance frequencies and to suppress the transmission of structure borne sound to neighboring speakers. Figure 8 shows a close-up of some loudspeakers combined to a wall.

7. SPATIAL AUDIO RENDERING SOFTWARE ARCHITECTURE

The previous sections described the realization of all hardware components, especially the audio network. The render units already have been introduced. They are intended to calculate the individual audio signals that feed the speakers. An additional PC is used to configure and parameterize the render units. The operating system of all involved computer systems is Linux. A real-time kernel is installed on the render units in order to achieve low-latency behavior. Like the modular audio network which allows different setups, the render units need to be flexible, too. The developed rendering software supports a seamless adaptation to different loudspeaker arrangements.

7.1. Rendering Framework

All render units use a flexible software framework to transform the audio input signals into audio output signals (rendering). The framework supports and utilizes multi-core hardware and provides low latency audio processing down to 2.6 ms (two times 64 samples at 48 kHz). It is based on data flow graphs composed of nodes, called blocks, to manipulate audio data. The audio blocks are connected by arcs, called connections, to transfer audio data between the blocks. A block usually represents a basic signal processing algorithm, e.g. a filter. One special block is used to access the system sound card(s) and allows reading and writing to them. There is a large library of predefined blocks including gains, equalizers, convolution algorithms, routing-matrices, delay-lines etc. up to complex spatial algorithms like wave field synthesis. The connections between the blocks, i.e. the arcs in the flow graph, are arbitrary. This concept is similar to the one used in the audio network, where network cables connect different modules.

A rendering algorithm, i.e. a specific set of connected blocks, is stored in a XML-file which is loaded by the software framework. Once started, the framework reads the data flow graph and automatically distributes the containing blocks to the available processor-cores to exploit parallel execution paths in the graph in order to achieve maximum performance.

Additionally the framework supports command connections via TCP and UDP to manipulate block parameters, e.g. gain values, at runtime. Theses connections are also defined in the XML-configuration-file. To transport arbitrary control data to the blocks, the OSC (Open Sound Control) protocol has proven to be useful [6].

The data flow paradigm combined with the rendering software allows an easy and fast development of signal processing algorithms. The rendering algorithms are only limited by imagination, because of the modularity provided by the underlying framework.

7.2. Loudspeaker Signal Routing

Both, the audio network and the software rendering framework are modular and flexible. Therefore, there are many possible combinations in mapping the logical channels provided by the render units to the physical loudspeakers.

One obvious and straight forward way is to use only one render unit to drive a subset of all available loudspeakers. An example is classic wave field synthesis on the circular array around the listening area or on a particular row of the loudspeaker wall. In this scenario the control-PC is used for interactive parameterization of the render unit.

To synthesize a real 3D wave field with the loudspeaker wall, multiple render units are necessary, because of the large number of output channels and the required computational performance. Those kinds of problems need to be divided into smaller subproblems, which are solvable by one render unit. Both, the audio network and the rendering software support and encourage this approach. In the explained setup each render unit uses its own configuration and is responsible for a specific subset of all loudspeakers, e.g. a few loudspeaker rows on the wall. The required coordination is provided by the control-PC.

8. APPLICATION

The presented massive multichannel loudspeaker system offers a large range of applications. As described in the previous sections, the audio network based hardware and also the software rendering environment is very flexible. The current arrangement of the multichannel loudspeaker system as a planar surface enables investigations into the synthesis of real 3D wavefronts emitted by virtual sources. This is currently done in a research project that focuses on the use of wave field synthesis in the virtual acoustic product development process. At the time the virtual product development process concentrates on visualisation of geometry and associated properties, to evaluate the properties of products, before real prototypes are build. In this context wave field synthesis enables the possibility to evaluate acoustic product properties. Figure 9 shows an example application where the user watches a stereoscopic visualization of a so called pick & place unit. The front projection is done on a perforated screen that is placed in front of the loudspeaker wall. A visual rendering software computes a continuous rotation of the virtual object around its vertical axis. Furthermore the information of position (including height reproduction) and rotation angle of the pick & place

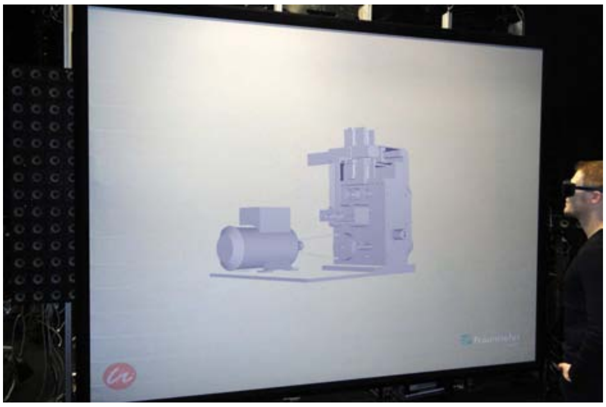


Figure 9: Use of the loudspeaker wall in combination with stereoscopic video projection. The images shows the visualization of a pick & place unit that is continuously rotated around the vertical axis. The render units that drive the loudspeakers behind the perforated screen generate signals for the auralisation of the directional sound field of the example machine.

unit is sent to the render units using the OSC protocol. Based on the knowledge of the loudspeaker and the virtual source position, the render units compute the audio signals of the loudspeakers. This includes the auralisation of the directivity of the virtual source, which will be updated in real-time for each new position and rotation angle of the virtual source. The system is installed in a special acoustic laboratory that was built according to the recommendation ITU-R BS.1116, which allows hearing tests in an ideal listening environment [11].

9. CONCLUSION

In this paper the concept, design and realization of a massive multichannel loudspeaker system for the use in spatial audio research is presented. Starting with basic requirements for such a system all components and design decisions of the system are described. One of the basic requirements was the ability to use all 640 channels of the system simultaneously as well as the ability to split the system into multiple independent units. This demands new concepts regarding audio distribution, wiring and administration. The foundation of the entire system is an audio network. To realize such a distribution concept common network topologies were discussed regarding their applicability. A daisy chain topology with redundancy was chosen. The used audio network is capable of transmitting 64 channels over one standard CAT5 cable at a sample rate of 48 kHz. This basic building block is repeated up to the requested number of output channels. Furthermore, the audio network protocol allows to transfer meta data to the connected network components. With this feature an adjustment of each audio channel is possible, e. g. for individual equalization and amplification.

All hardware components of the audio network are described, with a particular focus on a dedicated setup of the entire system, that is used for current research activities. The audio network is managed via a central interface called NCU which also feeds the MADI signals provided by consumer PCs equipped with professional sound cards into the grid of amplifiers. The developed 16 channel, digital, fan-less amplifiers are small enough to be mounted very close to the loudspeakers. The amplified signals are transmitted to the loudspeakers using standard CAT5 cables. This unusual cable choice reduces costs and weight of the entire system significantly. The employed loudspeakers achieve a good trade-off between size, weight and sonic quality.

Beside the audio network and its components, the very flexible, spatial audio rendering environment is presented as well. It is based on a similar distribution concept as the audio network. Here different audio signal processing blocks can be connected to form complex processing graphs.

As example, an application consisting of a stereoscopic video projection combined with the mentioned dedicated setup of a planar loudspeaker distribution is given.

10. ACKNOWLEDGEMENT

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