Active Array Loudspeaker

SoundFocus ApS

2011-01-17 Søren H. Nielsen

1 Overview

An active array loudspeaker is proposed which can radiate sound in a narrow beam in a selectable direction in order to provide individualised sound to one or more listeners.

A summary of desired functions and properties of the loudspeaker (more details in subsequent sections):

- focussed presentation of sound
- enabling/not blocking personal communication
- expected distance to listener(s): 0.5 to appr. 3 m
- direction (and beam-width) of sound from loudspeaker can be changed
- sound radiation can be minimised at desired direction(s)
- remote user interface
- analogue or digital signal input, at f_s=44–48 kHz
- possibly a network interface for signal and control
- 2 or more input channels, with individually settable direction
- 8-32 output amplifier channels
- playback level adapts to surrounding noise
- playback dynamic range adapts to surrounding noise
- power supply from mains or DC network
- low power consumption, both when idle and playing
- simple installation

The loudspeaker presented here is designed to be used in a context described in Nielsen [2011]. Figure 1 shows an overall block diagram of the loudspeaker using just 8 loudspeaker drivers.

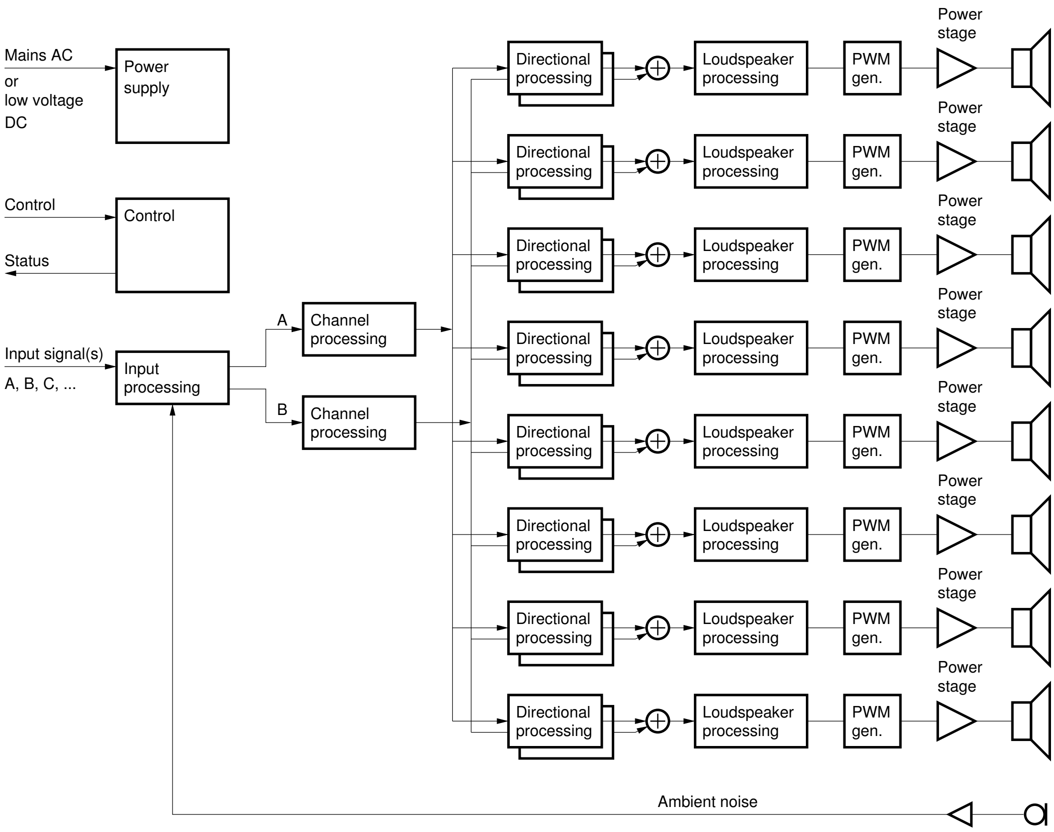

Figure 1: Overall block diagram of the active array loudspeaker – with 8 loudspeaker channels.

2 Processing

The signal processing is split into several stages as some of it is common for all input channels, whereas other parts are different for each output channel (loudspeaker driver).

A low processing delay is desirable in order to ease integration with other media, in particular moving pictures.

2.1 Common processing

A large part of the processing in the array loudspeaker is common for all loudspeaker drivers. Most of it is identical for several input channels, i.e. with identical parameters but independent signal paths.

2.1.1 Network packet processing

In case a network connection is used, the audio data received may be packed in a way that requires some processing, e.g. buffering and error concealment.

2.1.2 Decoding of encoded audio

As an option, the audio input can be in the datarate-reduced form (e.g. MPEG). DSP implementations of decoders can be obtained from a third-party.

2.1.3 Adjustment of overall frequency response

The loudspeaker driver units are small in order to provide good directional control over a wide frequency range. This has the unwanted side effect that the frequency response is likely to need correction in order to provide a good tonal balance.

2.1.4 Automatic adjustment of level according to surrounding noise

In some applications it is desirable to adapt the playback level to the general noise level at the playback site. In order to facilitate that, a microphone is used together with a suitable signal processing algorithm.

2.1.5 Dynamics processing

Dynamics processing may be needed for several reasons: First, it can be used to minimise the need for manual correction of level, depending of the nature of the source material. Second, it can be used to adjust the overall dynamic range of the source material to the conditions at the playback site (where backgroound noise may decrease the usable dynamic range). Third, suitable dynamics processing can increase the intelligibility of the source material.

2.2 Separate processing per output channel

2.2.1 Directional processing

The directional control comprises primarily settings the individual delays of the loudspeaker signals. The delays are up to appr. 3 ms and need to be settable with single-sample resolution or better (f_s=44–48 kHz).

In order to provide finer control of the directivity some filtering is also needed. Typically, both magnitude and phase need adjustment.

Note that the input signals may be given different directional processing, thus enabling different signals to be played in different directions.

2.2.2 Loudspeaker processing

In order to protect the loudspeaker driver and its power stage from hard clipping, some sort of limiter (dynamics processing) is desirable.

2.2.3 Generation of PWM signals

The conversion from PCM sample representation to pulse-width modulation for the power stage can be done either on-chip (by programmable logic or DSP) or by an external chip. SoundFocus has access to suitable IP to generate PWM signals using an FPGA.

3 Interfaces

3.1 Analogue input

For a small setup, the easiest audio input interface to use is analogue, in a quality corresponding to 16-bit quantisation or better. Numerous solutions for providing an analogue audio input exist.

3.2 Digital input

Digital audio input may be of a streaming or packetised type.

Some signal sources may provide their signal using an industry-standard streaming interface, such as AES3 or IEC985, and this interface is good for testing purpuses.

For a setup using several signal sources and array loudspeakers some sort of network is desirable. The three most important types in the present context is: USB (not really a network but still useful), wired or wireless Ethernet (IEEE802), and FireWire (IEEE1394). In some cases also Bluetooth may be a viable option.

3.3 Configuration and user interface

As the loudspeaker is typically placed out of reach all user interface most be remote, using either a dedicated connection or one of the above-mentioned networks. It is also desirable that configuration can be made on-site, i.e. without having to remove the loudspeaker from its playback position.

4 Power Amplifier Output Stage

Several commercially available chips providing PWM generation and output stage exist. In the case where PWM generation is done in our own hardware, just an output stage is required, for which also suitable components exist. As the output power requirements are moderate, cooling can be done on the PCB itself.

5 Power Supply

The power needed for the signal processing and interface is low and relatively constant. The power consumption of the amplifier output stages is varying over time when playing music, and may be in the order of 0.1–30 W in total. Different supply voltages for logic and power is needed.

One way of providing power is using the AC mains together with a suitable power supply.

For simlicity of installation it may be desirable to use the digital data network also to supply the power for active loudspeaker. Power over Ethernet, for example, offers up to 15 W in the basic configuration.

6 Technical platform

The platform for the active loudspeaker is obviously digital, but the choice of processing and interface platform is open. It may be a SoPC (System on Programmable Chip), a DSP, a general purpose microprocessor, dedicated chips, or a combination of these.

References

Søren H. Nielsen. Distributed sound system with directional control. Technical report, SoundFocus ApS, January 2011. Unpublished.