Error-Analysis and Optimization of Distributed Microphone-Arrays for Advanced Multichannel Signal-Processing

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Abstract

Real life applications in large scale audio-monitoring scenarios such as smart cities require cost-efficient high-performance audio-hardware that can be used on existing data infrastructure. Advanced multichannel signal-processing applications, for example beamforming and localization algorithms, require isosynchronous input signals with low channel interference and low correlated noise for all channels.
In this paper, the influence of hardware-designs for distributed microphone arrays on the signal-quality is examined. The most important characteristics for multichannel signal processing applications are signal to noise ratio, channel crosstalk, correlated noise, word-clock jitter, and signal latency. Sources of electrical interferences, correlated noise and corresponding error-paths in the analog front-end are identified and analyzed using error-models. Hardware optimizations for low noise and low correlation based on these findings are proposed.
A novel distributed microphone-array based on Field Programmable Gate Arrays (FPGA), UDP/IP-transport and -word-clock distribution has been developed and implemented in a single-device to enable direct multichannel audio-streaming within a network. The performance of the system for multichannel algorithms is compared with commercially available audio capturing devices with focus on localization and beamforming.

Introduction

Application of Acoustic multichannel signal processing schemes in real-world setups require reliable hardware interfaces. Especially for large-scale audio monitoring scenarios, e.g. for sensors installed in smart cities it is still a challenging task to properly design microphone array installations. Often this is accomplished using conventional off-the-shelf studio equipment, which is costly and time-consuming in installation and often does not meet the required form-factors for integration.
Arising problems for multichannel signal processing algorithms are:

Synchronization between microphone arrays is done on the system scale, including arrays and network bridges.

Latency from array to receiver is determined by the used network, microphone array and receiver.

Correlated noise occurs on the scale of a single microphone array.

In this paper we focus on the correlated noise arising in a single array, it's sources and possibilities to overcome it.

Microphone Array Design

The proposed Synchronous Recording Interface for Multichannel Processing (SHRIMP) - consists of multiple microphone-arrays that operate in a distinct UDP/IP-subnetwork [1]. The wordclock synchronization, necessary for isosynchronous input signals, is realized based on the same subnetwork using industry-standard realtime protocols (e.g. PTP [4]).
The structure of a single microphone array is shown in Fig. 1. The network connection is established using an embedded circuit based on a field-programmable gate-array (FPGA) which realizes the ethernet interface. Here, a finite-state-machine handles clock synchronization, audio interfacing, buffering and streaming.
The analog front end (AFE) and the analog-to-digital converters (ADC) are located on a custom printed circuit board (PCB).

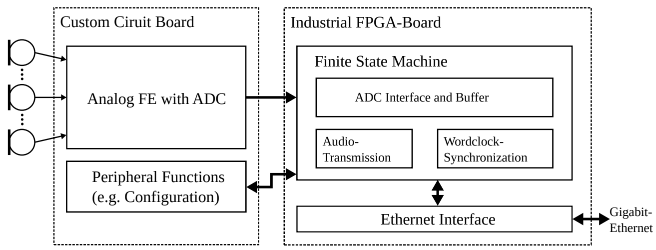


Fig. 1: Structure of single integrated microphone array.

The AFE provides 10-48V phantom power to the 10 integrated microphones. The microphones are connected using differential signaling to suppress irradiation of electromagnetic noise on the microphone cables.
On the circuit board holding the AFE, analog and other peripheral digital circuitry is located. This includes a serial interface for configuration of the microphone array, a persistent configuration memory and power supplies for phantom power, amplifiers and ADCs. These peripheral circuitry poses a potential source of Electromagnetic Emission (EME) coupling into audio signal paths.

Sources of correlated noise

The design-requirements of an analogue front-end for multichannel localization-hardware differs slightly from single-channel AFE design. Eliminating crosstalk and correlated noise is crucial, as parasitic signals common on all channels will decrease the accuracy of localization. While eliminating crosstalk means to enhance channel separation, the various sources of correlated noise are harder to eliminate. For single-channel applications, a very high signal-to-noise ratio is demanded. For localization, more noise can be tolerated, if it is uncorrelated among the input-channels. The identified dominating noise sources of correlated noise are (see Fig. 2)

- External noise introducing error via electromagnetic emission (EMI) on cables and feedlines, $V_{noise,EMI}$. Coupling m_n is depended on geometry and distance to noise-source.
- Noisy bias-source on single-ended lines, resulting from noise on common analog supply, $V_{noise,VBIAS}$
- Noisy supply or common ground-planes on active pre-ADC filters, $V_{noise,VCC}$
- Insufficiently filtered and buffered supply-voltages of the ADC, $V_{noise,VDD}$

In practical applications, the microphones are placed in arrays of various, often sparse geometries. To allow the suppression of external noise coupled into the cabling requires a careful design of the symmetric (balanced) lines of each channel to keep the error common on both lines. To eliminate this error, a high common-mode rejection-ratio (CMRR) is needed. Thus, a balanced, DC-controlled instrumentation-amplifier was selected as the main input amplifier of the SHRIMP, rather than relying on the (generally worse) CMRR of a symmetric ADC. High-precision passive components are mandatory in the balanced path.
When choosing a single-ended (non-symmetrical) ADC, other parts of the design are sensitive to internal sources of correlated error. The filtering of the mandatory biasing voltage is crucial, as noise from the common analog power supply is fed directly to the input-signal of all channels. Here, an individual voltage-reference (diodes) and buffers are used for each channel to separate the offset-voltages. Pre-ADC filtering should incorporate amplifiers with a good power-supply rejection-ratio (PSRR) and the layout should consider individual ground-planes for each channel to minimize the error due to ground-pumping.
Using balanced ADCs allows a more relaxed design of the filter stage, but requires another set of amplifiers for balancing the output of the main instrumentation amplifier.
In of-the-shelf dual-channel ADCs, some coupling (crosstalk) between the channels of each ADC can always be observed. This can only be avoided when using single-channel ADC. To minimize correlated noise, analog power supplies must be buffered and filtered thoroughly, as the PSRR of the ADC is generally poor.

The AFE-design in the SHRIMP shows good CMRR performance for external noise-sources, but it can be observed that common-mode noise is introduced within the device, especially on those channels placed closest to the high-current digital processing units. A revised version should investigate if high-CMRR instrumentation amplifiers and balanced analog-to-digital converters can reduce the internally correlated noise further.

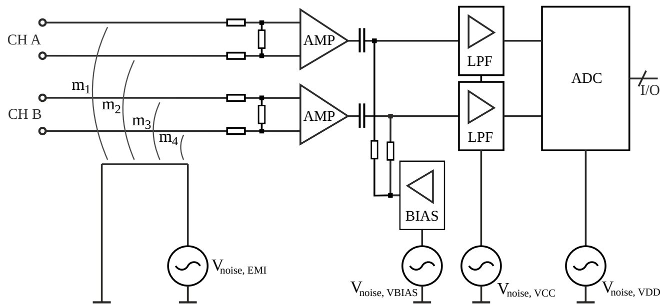


Fig. 2: Sources of correlated noise in multichannel analog front-ends (AFE), one ADC (two channels) shown.

Correlation analysis

The correlation between individual channels is measured using the magnitude squared coherence (MSC) [2]:

$$C_{xy}(f) = \frac{|G_{xy}(f)|}{G_{xx}(f)G_{yy}(f)} \tag{1}$$

where f is the frequency, G_{xy} is the cross-correlation of the two signals and G_{xx},G_{yy} are the autocorrelations of the input signals respectively.
It shows the normalized correlation between the two signals.

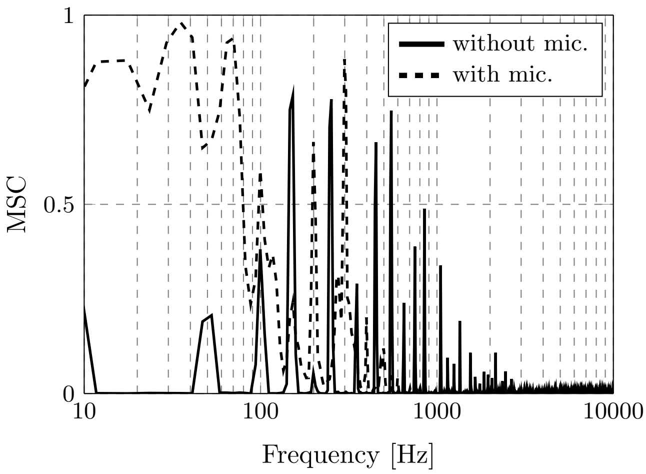
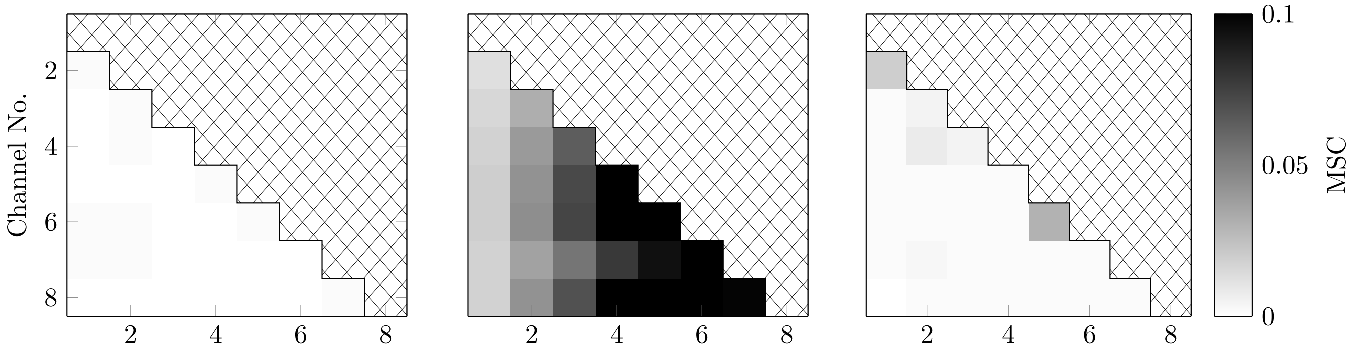


Fig. 3: MSC of Behringer ADA8000

The MSC of two channels of an AFE is strongly dependent on the total analog gain of the AFEs. It can be observed, that on some devices, an increased gain setting corresponds to lower MSC. This can be explained by an increase of uncorrelated noise from the analog circuitry



(a) RME Octamic II. (b) Behringer ADA8000. (c) SHRIMP.

Fig. 4: Mean MSC of channel-combinations of all three test subjects.

(thermal noise) that masks the common noise from digital components and, thus, reduces correlation.
To compare different AFEs, the gain settings have to be matched. A fixed voltage gain of 10dB for all three AFEs under test was chosen.
In figure 3 two MSCs of the same channel combination are shown. One with attached microphones, measured in an anechoic chamber, the other one measured without any microphones attached. It can be observed that the MSC in the measurement with microphones is high at lower frequencies. This originates from ambient noise not damped by the anechoic chamber due to the lower cutoff frequency and is not a consequence of electromagnetic emission in the AFE.
With no microphones are attached, the only sources for correlated noise are the internal sources of the AFE and other circuitry in the device.
Since we want to focus on exactly these internal sources, all subsequent measurements are taken without microphones attached.
To visualize the MSC of different AFEs, the mean MSC was calculated over a frequency range of 150Hz to 5kHz of all channel combinations and is shown in figures 4a to 4c. For the comparison, two commercially available sound interfaces in different price ranges are chosen.
The *RME Octamic II* [5] shows very low MSC over all frequencies and channel combinations. The AFE utilizes a fully differential signal path from microphone input up to the ADC, which by itself is differential.
The *Behringer ADA8000* [6] shows relatively high MSC especially between channels with higher channel numbers. The ADCs for channels with higher channel number are placed closer to digital circuitry and the power supply on the PCB. The AFE provides differential inputs, but single-ended ADCs, so the signal is single ended after first amplification.
Our system, the *SHRIMP*, shows higher mean MSC than the RME but lower than the Behringer AFE. It uses the same AFE topology as the Behringer, differential input and single-ended ADC.

Conclusion

An integrated microphone array of which multiple arrays was set up that can be operated in a UDP/IP network. This development includes an analog front end. The performance in terms of correlated noise of this front end is comparable to commercially available audio interfaces.
The examined front ends with fully differential signaling paths from input to ADC generally show better performance. Thus, when low correlated noise is essential one should choose this front end topology. However, if single-ended ADCs - consequently partly sigle-ended signal paths - are chosen, internal analog and digital noise sources should be spacially seperated or shielded to prevent electromagnetic emission to the analog signal paths.

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