24th INTERNATIONAL CONGRESS ON SOUND AND VIBRATION
23–27 July 2017, London

ACOUSTIC INTERNET-OF-THINGS: LOW-COST, DISTRIBUTED, SYNCHRONOUS MEASUREMENT SYSTEMS FOR AUTOMATED SOUND SOURCE LOCALIZATION IN WIND TURBINES

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This paper describes first results of the development of a distributed, synchronous acoustic measurement system, which combines lowest cost and highest robustness for automated sound source localization in wind energy parks and assessment of wind turbines acoustic signatures. After systematically collecting and structuring the requirements, a scalable system was developed based on subunits of multiple MEMS sensors, an FPGA, and a microprocessor. The system automatically synchronizes across multiple subunits and allows fully automatic, synchronous data collection over extended periods of time in harsh environments. The data collected is analysed and sound sources are identified based on beam-forming algorithms. Application of this method allows simple and automated identification and localization of sound sources on single wind turbines in wind parks. Based on a systematic assessment of the audio data collected a database of wind turbine sounds is created, which is intended as a base for the auralization of future wind parks and for automated pattern-based assessment of the condition of wind turbines.

Keywords: Wind Turbine Sound, IOT, Acoustic Camera, Auralization, Condition Monitoring

1. Motivation and Introduction

The large-scale industrialization of wind turbines, leading to an extensive growth in both number and size, has raised questions regarding potential health impacts on nearby residents due to noise emission [1]. However, sound emission from wind turbines is complex, variable over time and shows characteristics not commonly included in acoustic characterizations. A consistent, long-term assessment would be required [2], which is not yet feasible with state-of-the-art measurement equipment for localizing sources of noise emission, due to robustness and cost. Thus the development of means for the systematic, long-term observation of sound emission from wind turbines at a low price point enables a new level of quantification of emissions and can improve the overall acceptance of wind turbines. In addition, consistent monitoring devices and their integration into the wind park control can be used to minimize noise emissions by selectively switching single turbines off, while at the same time the over all on-time is kept at a maximum level.

The implementation and the results presented here, are based on a consistent product development methodology and include the first performance measurements of a low-cost, distributed acoustic measurement system, which indicate a significant potential of the architecture developed.

2. Methodology

The development of a low-cost acoustical camera for the monitoring wind turbines and integration into the Internet of Things (IoT) presented in this paper, is based on application the V-model for the development of mechatronic products [3].

The requirements of the application are systematically developed and mapped to the requirements of an acoustic camera, which is to be integrated into the nacelle or the tower of a wind turbine. From these, the technical requirements for microphone units are deduced. After completing the requirements for the acoustic sensors, the system architecture is proposed, which enables a powerful, scalable system at a very low price point. The system is physically implemented and first tests are performed and presented.

3. Development of requirements

Since the development of wind turbines is mostly driven by mechanical and power-electronics, hardly any standards are available specifying e.g. robustness requirements for sensors and measurement systems [4, 5, 6]. In this paper a basic set of requirements for this type of systems is proposed. Over-engineering is to be avoided, since the goal of our study is a low-cost system. The most relevant requirements for the acoustic measurement system are deduced based on a minimum approach backed by experimental data and physics modelling.

3.1 Technical requirements

For all acoustical measurements the frequency range of the observed signals is crucial with respect to the used microphones and any type of signal conditioning. Thus frequency spectra of wind turbines under variable conditions have been measured, analysed, and compared to literature [7, 8]. Figure 1 shows an example measured in a wind park close to Hamburg. The noise level is dropping below -90 dB for frequencies above 9 kHz, as shown in figure 1. Thus, based on the Nyquist theorem, a minimum sampling frequency of 18 kHz would be sufficient to properly measure the acoustic emission.

Figure 1: Distribution of frequencies of a typical wind turbine emission (author’s measurement)

To map the geometric resolution, which represents the minimum detectable distance between two independent sources, to time-based microphone parameters, a simplified calculation for a single microphone is performed, mapping the lateral resolution Δx to a minimum time interval ΔT, which must be properly separated within the measured data. The used parameters are indicated in figure 2.

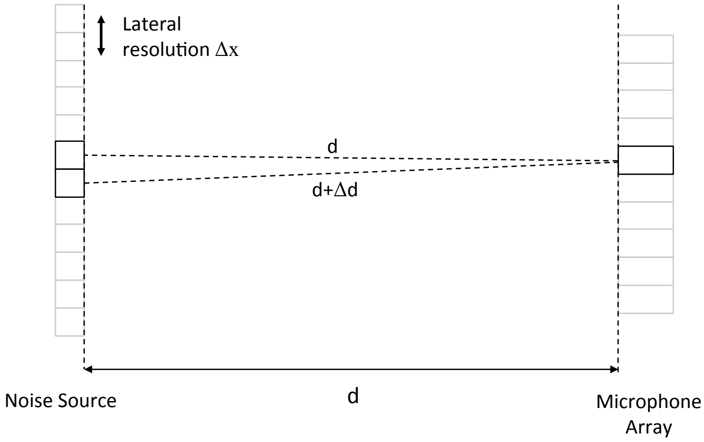

Figure 2: Simple model to map geometrical resolution requirements to timing requirements

Based on a distance of $d = 100$ m between the microphone array and the object measured and a lateral resolution of $\Delta x = 1$ m, simple geometry leads to $\Delta d = 5$ mm. With the speed of sound in air for a temperature of $\vartheta = 18$ °C of $c = 342$ m/s this leads to a time difference of $\Delta T = 15$ µs, that the measurement system would have to separate. This value corresponds to a minimum sampling frequency of $f_s = 68.4$ kHz.

Passive acoustic source localization at low sampling rates is addressed in current research, applying generalized cross-correlation in combination with up-sampling [9]. To obtain an estimation for the minimum required sampling rate in a microphone array, a simplified model is applied: If a second microphone is added in a geometrically defined position to the first one and both are synchronously sampled, the setup would actually allow a localization with the precision defined above reducing the sampling frequency for each single microphone to $f_s = 34.2$ kHz. With further microphones added to the setup, a further reduction of the minimum required sampling frequency is expected. Thus, assuming an array of 12 microphones, the sampling frequency of $f_s = 40$ kHz used in the setup due to data-rate limitations in the signal path, is expected to be sufficient to both fulfil the Nyquist theorem for the maximum noise frequency of 9 kHz (see figure 1) and at the same time fulfil the timing requirement for proper localization. Further technical requirements are summarized in table 1.

Table 1: Technical requirements.

Category	Requirement	Minimum	Maximum	Unit
Acoustics	Frequency Range	0.02	9	kHz
	Sampling Frequency	34.2		kHz
	Measurement Distance	50	300	m
	Lateral Resolution		1	m
	Number of channels	10	100	
	Distance between Microphone Units	0.01	3	m
	Distance between Branch Controllers	0.1	100	m
Electronics	Supply Voltage	4.75	5.25	V
	Supply Current per Channel (including branch controllers and supervisor unit)		50	mA
Environment	Temperature Range	-20	+50	°C
	Generator Vibrations	50	100	m/s^2
	Tower Vibrations	0	10	mm/s^2
	Humidity	0	100	%RH
	Wind	0	37	m/s
	Salt Spray (offshore application)			
Other	Total System Weight		10	kg

3.2 Non-Technical requirements

Besides the technical requirements several additional requirements were identified and taken into account:

- System cost: For integration into wind turbines with a significant installation rate cost becomes crucial. The goal for the study is a cost of 100 €/channel.
- Installation effort: Besides the cost of the system, minimum effort for the installation is a must. Significant reduction in the number of connectors and cables compared to existing solutions is expected resulting also in a reduction of setup time.

4. Implementation

In literature, designs including digital MEMS microphones, which are directly connected to an FPGA are described. The advantages are based on a fully digital signal path and the power of parallel, FPGA-based filtering [10]. However, these designs do not address the requirement to minimize installation effort.

4.1 System Implementation

Since the measurement system described here is supposed to be fully scalable a hierarchical architecture with a supervisor and synchronized, lower-level branch controllers is proposed. Figure 3 shows the block diagram of the proposed measurement system. The lowest level component is a digital MEMS microphone. Several of these are connected to a branch controller, which provides the interfaces for local data collection and time stamping. All branch controllers in a measurement system use a common clock to ensure proper synchronicity between the different branches. Every branch controller is connected to a supervisor unit, which performs the overall data collection and the interface to the Internet of Things providing remote access to the system and the measurement results. While microphones and branch controllers have to fulfil real-time requirements for synchronous data acquisition, collecting and conditioning of the data in the supervisor-unit can be performed asynchronously offline.

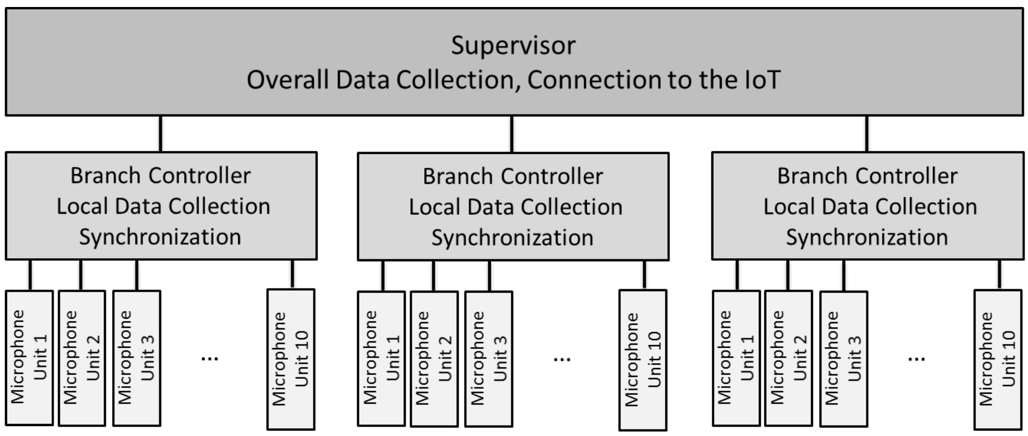

Figure 3: Hierarchical architecture of the measurement system

4.2 Component Implementation

In this subsection the implementation of microphone unit, branch controller and supervisor are described.

4.2.1 Microphone Unit

Several available MEMS microphones with digital interfaces were tested and an STM MP45DT02 [11] was chosen for the setup, since it fulfils all technical requirements and offers the option to manually solder. The microphone is connected to a microcontroller Atmel-SAM G53N19 [12], which is also part of the microphone unit. The processor includes a PDM-interface to the microphone and an SPI-interface to the branch controller. The block diagram of the microphone unit is shown in figure 4.

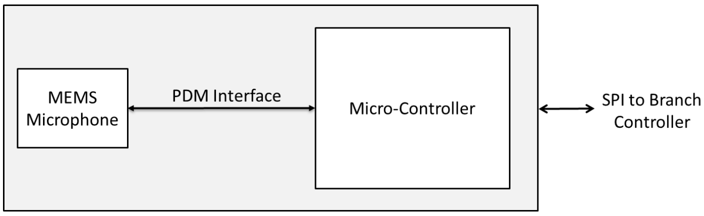

Figure 4: Block diagram of the microphone unit

4.2.2 Branch Controller

To implement the branch controller, a combination of a Xilinx Spartan 6 [13] and a single board computer was chosen. The FPGA is programmed to include up 10, fully parallel SPI interfaces, which is a must for synchronous measurements. The single board computer is running a UNIX-based operating system with extensions for precision timing and real-time operation. Figure 5 shows the block diagram of the branch controller.

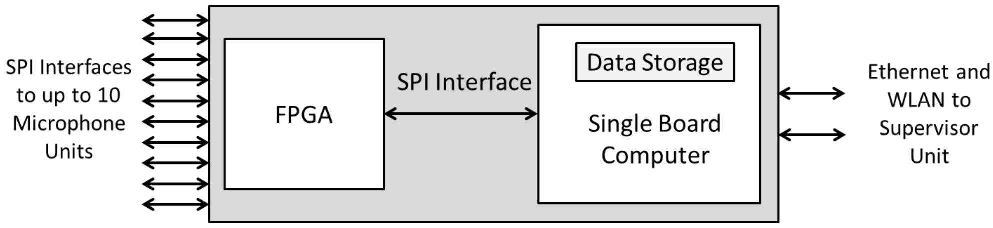

Figure 5: Block diagram of the branch controller

4.2.3 Supervisor

Since all real-time requirements are addressed within the microphone unit and the branch controller, standard Ethernet is used to gather the data of the branch controllers and connect the system to the Internet. The supervisor is based on a single board computer running a UNIX operating system (see figure 6).

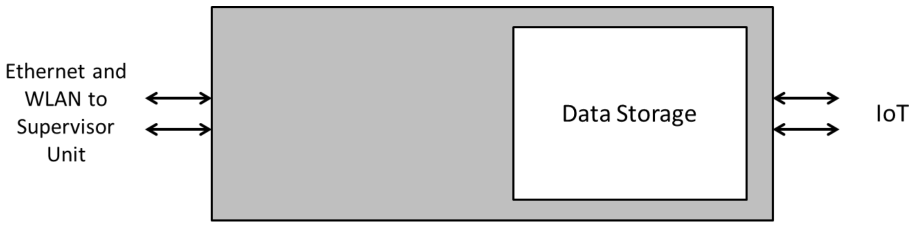

Figure 6: Block diagram of the supervisor

5. Measurement Results

5.1 Microphone Characterization

The output signal of a single microphone was measured using a calibrator at f = 1 kHz at 94 dB (figure 7). Besides the main peak at f = 1 kHz, additional maxima are visible in the spectrum, which are potentially caused by an amplitude modulation or aliasing effects. The measurement indicates a signal to noise ratio of about 40 dB, which is worse compared to the specified datasheet value of typically 61 dB [10].

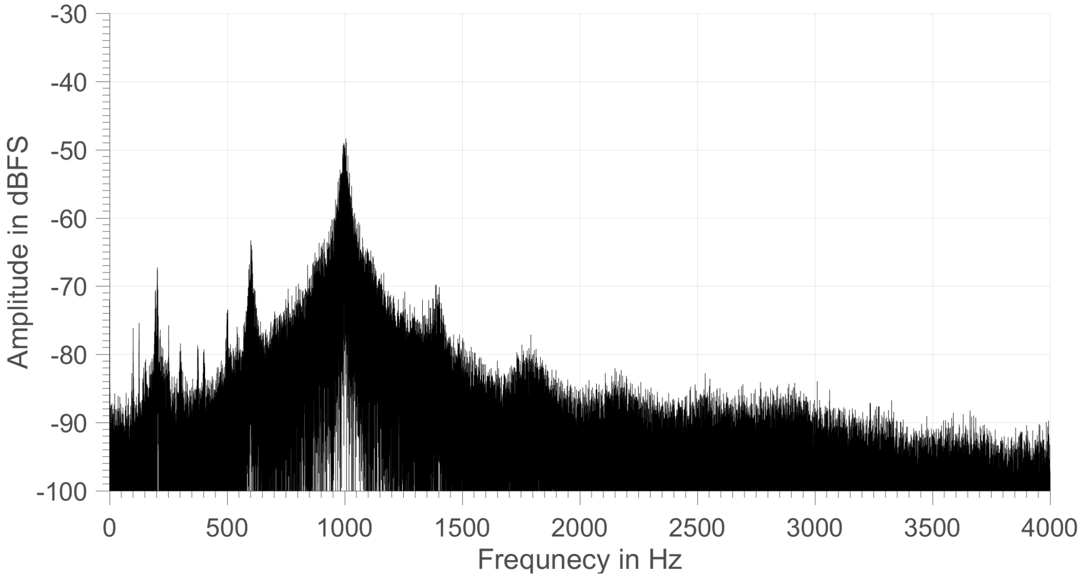

Figure 7: Characterization of a single microphone with a reference signal of 1kHz at 94dB

The frequency response was analysed up to 10 kHz with a simple loudspeaker setup (figure 8) using a slowly swept sine signal. The difference in amplitude of the Brüel&Kjaer 4190 reference microphone [14] and the digital microphone unit are displayed. In the range up to 9 kHz the deviations from a flat frequency response can easily be digitally corrected.

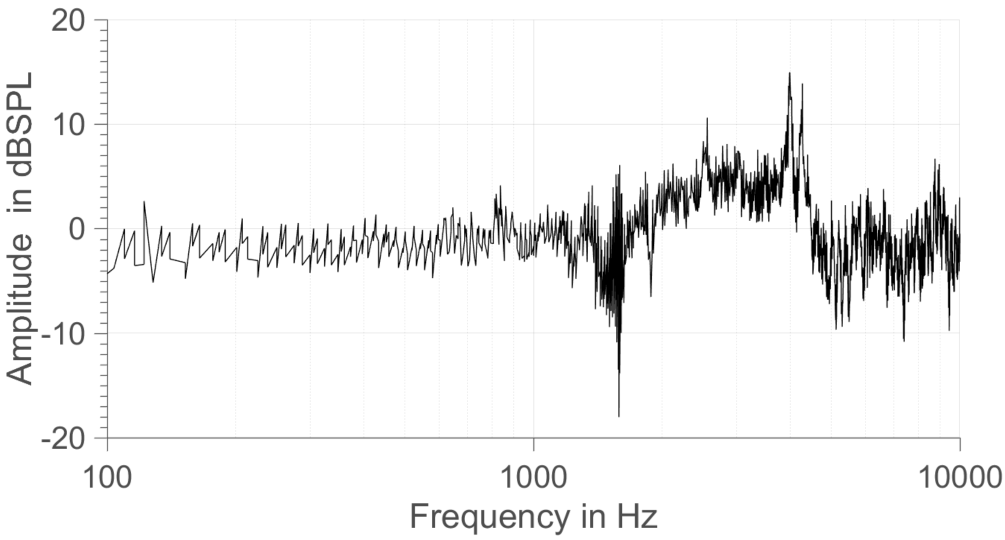

Figure 8: Characterization of the frequency response of one microphone unit with respect to a Brüel&Kjaer 4190 reference microphone

Finally, six microphone units have been linearly setup in equidistant positions with a spacing of 7 cm as indicated in figure 9a. The source was positioned at 45° and a distance of d = 2.5 m from the center of the array. Figure 9b shows the measurement result for a sinusoidal signal with an amplitude modulation. For two adjacent microphones a delay of ΔT = 140µs is measured, which is consistent with a calculation based on the geometry of the setup.

a) b)

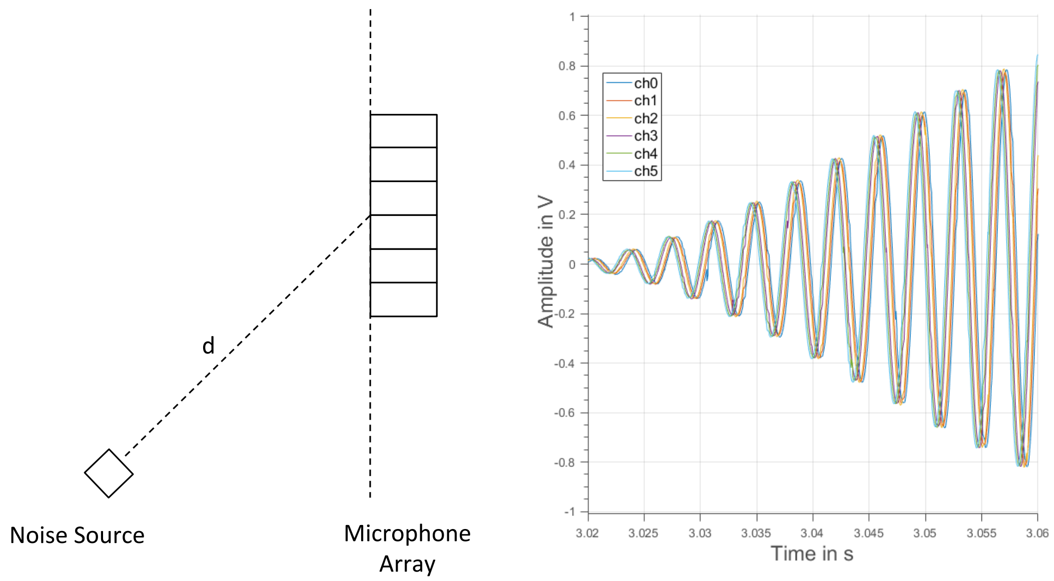

Figure 9: a) Acoustic setup for characterization of a branch of microphone units and b) measurement result

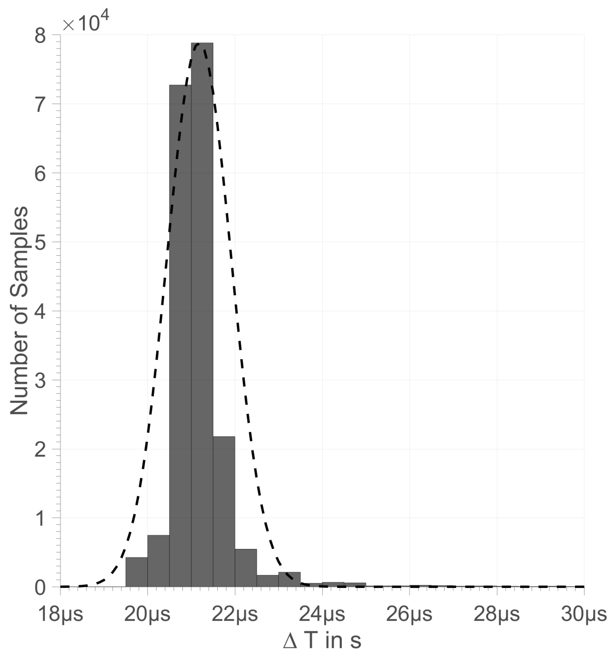

Figure 10: Measured synchronicity

To experimentally check the performance of the system, the timestamps of adjacent measurements have been compared and their difference has been statistically analysed for a total of 200,000 data points at a nominal sampling frequency of f_s = 48 kHz. The result is shown in figure 10, which indicates constant measurement intervals with a standard deviation of σ = 0.73 µs and an average value of μ = 21.18 μs. These results indicate a sufficient precision and rate of sampling points to achieve the respective technical requirements (table 1).

6. Conclusion

In this paper the first results of the development of a distributed, synchronous acoustic measurement system are presented. The scalable system combines lowest cost and highest robustness for automated sound source localization. The overall architecture is included as well as the block diagrams of the implemented subunits. First measurements of the system performance are presented and evaluated with respect to the requirements. The results clearly indicate the potential of structure integrated MEMS sensors for automated sound source localization in wind parks. Further development and measurements are planned to fully integrate the measurement system and additionally include sensors for the infrasound range.

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