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CIRA ACOUSTIC ANTENNA REAL-TIME FPGA BEAMFORMING ALGORITHM VALIDATION

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ABSTRACT

In the program LOWNOISE, funded by the Italian Ministero dell'Istruzione, dell'Università e della Ricerca (MIUR), the Italian Aerospace Research Centre (CIRA) was in charge of developing an Acoustic Antennas System to detect and characterize cars emitted environmental noise. A fundamental part of the Antennas System was the acoustic beam-forming algorithm, implemented on FPGA platform to satisfy real-time constraints and numerically tested to verify its feasibility.

In this paper, a test strategy for the numerical evaluation of beamforming algorithms is presented. Characteristic microphone signals, representative of physical acoustic antenna noise measurements, were defined and generated by using the Matlab Phased Array toolbox. In the first part of the paper, test signal description used in the evaluation is shown.

In the second part, algorithm's performance in terms of numerical accuracy, latency and FPGA resource usage (from the hardware point of view) and main lobe amplitude and Maximum Side-lobe Level (from the software point of view) are reported as outcomes of the numerical test campaign performed. The algorithm shows good processing capabilities and is able to efficiently attenuate interfering noise sources and to reject background noise. The spatial accuracy is found as expected from theory.

1 INTRODUCTION

In recent times, an increasing interest on microphone phased arrays has appeared both in technology and in literature. They are used for spatial filtering of acoustic signals coming from one or more sources, and to track sources' movements by means of triangulation. A lot of possible applications can be envisaged (from drone detection and tracking to environmental noise measurements -aircraft take-off and landing, train and car pass-by noise-, etc). Microphones phased arrays are passive systems (they can detect noise sources without emitting any signal) and are cheaper with respect to traditional systems based on electromagnetic signals (RADAR). They benefit of the improvements in the SmartPhone market, since they can utilize the same MEMS microphones that are used in COTS SmartPhones.

Signals coming from the microphones can be processed in a variety of ways. Because of the large amounts of data, processing such data in real-time is not trivial. Recent efforts have focused on using hardware-programmable devices such as Field Programmable Gate Arrays (FPGA), which are renowned for their parallelization potential, in order to attain real-time throughput from data coming at high speed (high levels of bit quantization and sampling frequency).

The system object of the present work has been developed inside the Italian Ministry of Education, Universities and Research (MIUR) funded project LOWNOISE. The aim was to build a system based on microphone phased arrays to be used in pass-by-noise tests to track an auto vehicle and, contemporarily, characterize its environmental acoustic emission in real-time. The microphone phased array was first designed by CIRA in [1] resulting in a system with 80 optimized randomly placed microphones, with acoustic signals quantized at 24 bit and sampled at 96 kHz. The Beamforming algorithm was designed and FPGA-implemented by CIRA in collaboration with the University of Naples Parthenope. Before manufacturing the overall system, the beamformer was tested with synthesized signal in order to verify its features.

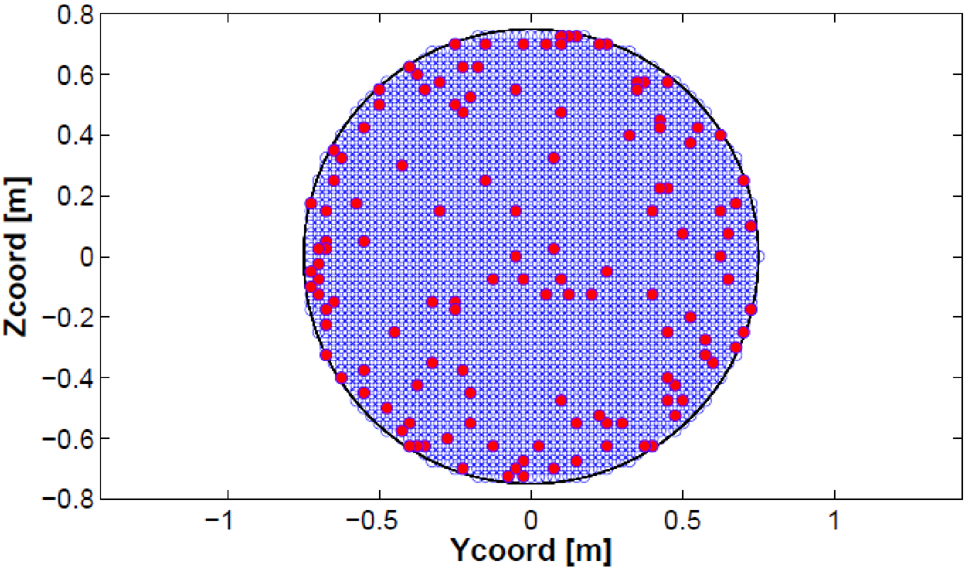


Fig. 1. Optimized random antenna designed by CIRA inside Low Noise project.

In the present work, we focus on numerically testing the hardware-implemented (FPGA-implemented) real-time beamforming algorithm to verify that the expected performance is in-line with that envisaged in the design phase. In the first two sections, a brief description of the selected beamforming algorithm and its FPGA implementation are briefly recalled. Following sections describe the numerical test strategy planned to verify the FPGA-implemented beamforming algorithm and then the results of these tests. Tests results are briefly commented in the Conclusions section.

2 REAL-TIME FPGA BEAMFORMING ALGORITHM

Beamforming is a technique to perform spatial filtering of signals (in the case of the present paper, acoustic signals). This technique applies to signals acquired by means of an array of sensors (microphone sensors in our case). A beamforming algorithm allows steering digitally the sensor array in a particular direction in order to add coherently the acquired sensor signals as they come from that direction. In this way, signals coming from the pointing direction are reinforced, and the other coming from other directions are faded away. To perform the calculation, an estimation of the time of arrivals of the signal from the pointing direction to the different sensors is necessary, as reported in the following picture

The beamforming algorithm implemented for the present work is a filter and sum Linearly Constrained Minimum Variance (LCMV) Beamformer (see Fig. 2). Acoustic signals are first filtered with a FIR filter with weights w_i. They are then properly delayed (array steering) and at last summed up. LCMV allows minimizing the harmful effects caused by noise, by interfering sources, or by the low frequencies content of the acquired signal.

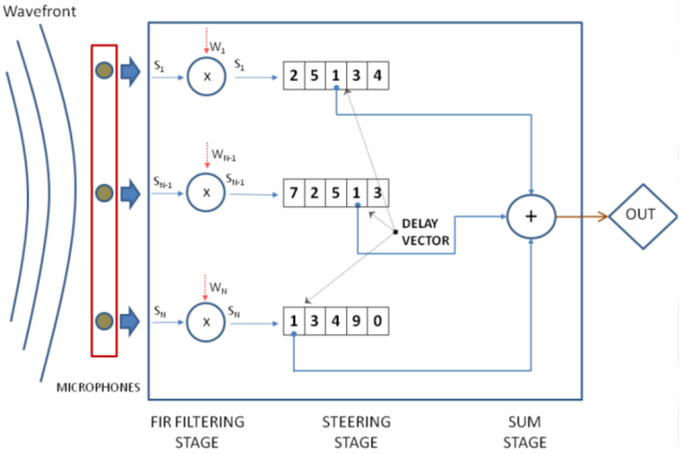


Fig. 2. Filter and Sum beamforming: signal acquisition, filtering, delay and sum stages.

The filters' coefficients w_i can be determined by means of:

- The signal spatial covariance matrix $\boldsymbol{R_y}$;
- A constraint matrix $\boldsymbol{C}$: the constraints allow imposing a null response from a certain steering direction and symmetric coefficients for each of the filters (in order to result in a Linear Phase beamformer);
- A response vector $\boldsymbol{c}$ such that $\boldsymbol{C'w} = \boldsymbol{c}$

A simple approximation of the covariance matrix is given by the following formula:

$$R_y = \frac{yy^H}{s} \tag{1}$$

Where y is a matrix with N columns, as many as the number of microphones of the array, and s rows, as many as the number of samples for each acquired microphone signal. Using the LCMV algorithm, the filter coefficients can be calculated with the following formula

$$w = R_y^{-1} C \left(C' R_y^{-1} C \right)^{-1} c \qquad (2)$$

The algorithm minimizes the power of the interfering signal (the background signal) $p = \boldsymbol{w}' \boldsymbol{R}_y \boldsymbol{w}$ subject to the constraint $\boldsymbol{C}' \boldsymbol{w} = \boldsymbol{c}$.

3 REAL-TIME BEAMFORMING: IMPLEMENTATION ON FPGA

An FPGA board is structured as an array of logical gates, similarly to common VLSI integrated circuits. However, FPGAs are not constrained to a single configuration, and can be freely re-programmed. Therefore, an algorithm designed for FPGA can be completely parallelized. Advantages of the FPGA architecture are:

- great computational capacity
- low power consumption
- zero overall dimensions

The classic FPGA programming method involves the use of hardware modeling languages (HDL or Verilog). In the last years Xilinx has issued the Vivado HLS (High-Level Synthesis) development environment, which allows FPGA programming starting from C code in a simple way. HLS allows realizing a top-level function in C language, which will be translated into a hardware module, and whose arguments represent the interface to the module itself. The hardware module is then ready to be deployed to a FPGA board. In our study we have used the Virtex 7 series of the Xilinx manufacturer. The beamforming algorithm implemented on the FPGA is described by the block diagram in Fig. 3

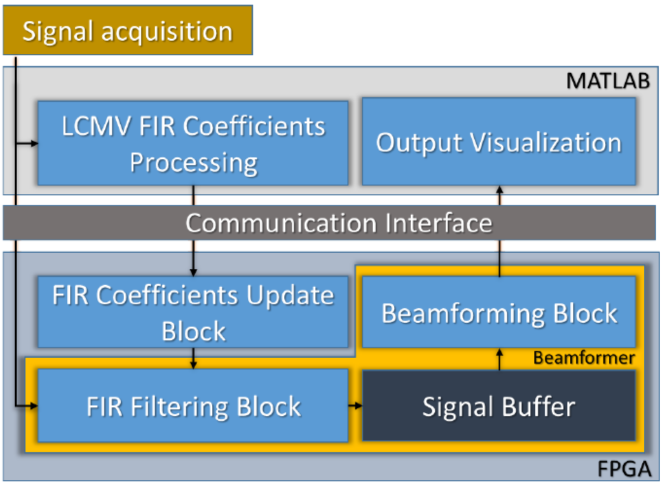
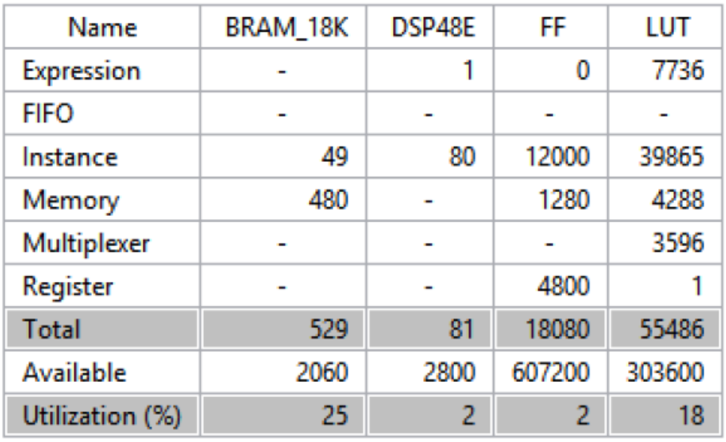


Fig. 3. FPGA implementation of the Beamforming system developed at CIRA.

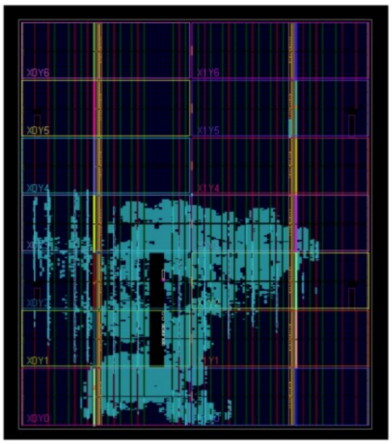
The weights for the filtering stage of the FPGA filter-and-sum beamformer are calculated by a function implemented in the Matlab environment that performs LCMV analysis of the synthesized signals representing the background noise without any moving sound sources. Repeating the analysis stage, thus updating the FIR weights, might be necessary from time to time in highly dynamic and noisy environments. The beamformed signal obtained by the FPGA is post-processed in Matlab as well for visualization purposes. The FPGA part firstly performs filtering of the 80 microphone signals, then stores the filtered signals in a buffer in order to reduce the computational time of the FPGA (increasing throughput). Finally, the signals are delayed and summed to obtain the beamformed signal for the different pointing directions (specifically, the "sound power" coming from the pointing directions). In [7] we discuss how our algorithm works in further detail.

FPGA resource usage to implement our algorithm is given in the table below: as it can be seen, only a limited part of the available resources is used, and these are mainly memory resources to store the data necessary for computation.

Table 1. XILINX Virtex 7 FPGA: resource usage.



Name	BRAM_18K	DSP48E	FF	LUT
Expression	-	1	0	7736
FIFO	-	-	-	-
Instance	49	80	12000	39865
Memory	480	-	1280	4288
Multiplexer	-	-	-	3596
Register	-	-	4800	1
Total	529	81	18080	55486
Available	2060	2800	607200	303600
Utilization (%)	25	2	2	18



4 NUMERICAL TEST STRATEGY

In order to evaluate the developed beamformer, different test signals have been generated through the Matlab Phased Array Toolbox making use of the following Matlab objects:

- OmnidirectionalMicrophoneElement: to represent the sensor elements;
- ConformalArray: to represent the array. Microphone number and position, Sampling frequency, Speed of Sound and the other parameters affecting the data acquisition process can be modified inside this object;
- WidebandCollector: to represent the array data acquisition process. Starting from a source signal, signals as read by each microphone can be easily recovered.

For each test case, these steps have been followed to generate the array test signals:

- Definition of the Microphones Array: by using Matlab objects, according to the selected characteristics of the single microphones and to the geometry of the array
- Noise signal generation with arbitrary typology and frequency: definition of signal parameters (frequency content, direction of propagation with respect to the microphones array, speed of sound and sampling rate);
- Generation of the simulated measured signals from the transducers array by means of the previously mentioned Matlab objects;

- Gaussian and interference noise (if required) is added to the signals, with its amplitude depending on the sampled signal (the noise added is different for each microphone);
- Signal conversion in 24-bit PCM format and scaling to use entire quantization range;
- Signal sending and elaboration to the Beamforming FPGA module by means of suitably formatted packets;
- Processing and reading of the acoustic map;
- Scaling (respect to the square of the input scale factor) to get the original power values;
- Displaying the generated acoustic map.

A total of 160 test cases have been analyzed to test the hardware-implemented beamforming algorithm. Each test case differs for the following parameters, characterizing the test scenarios:

- Type of signal: Sine signals with several frequency (500, 1250, 2000Hz) and Chirp signals (sweep from 500 to 2000 Hz);
- Added noise amplitude: 5, 25, 50, 75 e 100% with respect to the signal amplitude;
- Noise interference present or not, at the same frequency of the clean signal and at 50% of its power, coming from different directions;
- Propagation direction coincident or not with a pixel of the acoustic map;
- Double-precision floating-point or fixed-point numeric types (to evaluate the performance of FPGA implementation -that natively uses fixed-point arithmetic- with respect to CPU implementation, which makes use of floating point arithmetic-).

5 NUMERICAL TEST CAMPAIGN & RESULTS

In the following paragraphs, the test campaign results are presented to compare the effect of the various parameters affecting the beamformer's performance:

5.1 Fixed-point vs Floating-point arithmetic effect

The first result we present is the difference in accuracy of fixed-point arithmetic against floating-point, in Table 2. In fact, floating-point arithmetic is often not suitable for use in FPGA algorithms, because of the latency and resources required in order to perform floating-point operations.

Table 2. Acoustic map max error: Fixed-point vs Floating-point arithmetic.

Error type	**Mean Error** [dB or %]	**Std Error** [dB or %]
Absolute (500Hz - 1250Hz - 2000Hz)	0.023	0.089
Relative (500Hz - 1250Hz - 2000Hz)	0.001	0.009
Absolute (500Hz)	0.008	0.031
Relative (500Hz)	0.001	0.015
Absolute (1250Hz)	0.024	0.096
Relative (1250Hz)	0.001	0.005
Absolute (2000Hz)	0.037	0.139
Relative (2000Hz)	0.002	0.007

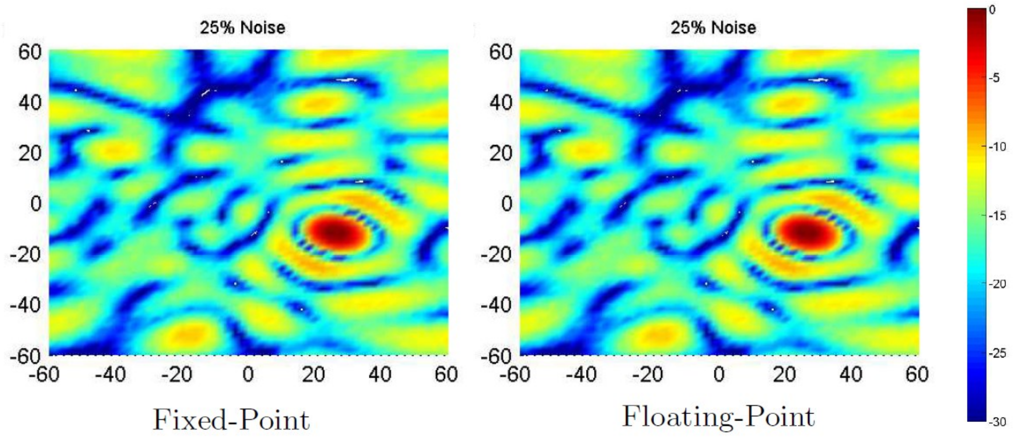


Fig. 4. Fixed-point vs Floating-point acoustic maps.

The differences in accuracy using the fixed-point arithmetic with 24 bits of quantization are negligible with respect to using floating-point arithmetic, and they do not result in meaningfully different acoustic maps.

5.2 Source signal frequency content effect

With respect to the frequency content of the source signal, we obtain the maps presented in Fig. 5 when passing from an acoustic source of 500Hz, to an acoustic source of 2000Hz. The values of the Maximum Sidelobe Level and of the Main Lobe Amplitude are coherent with the values that we expect from the theory [8].

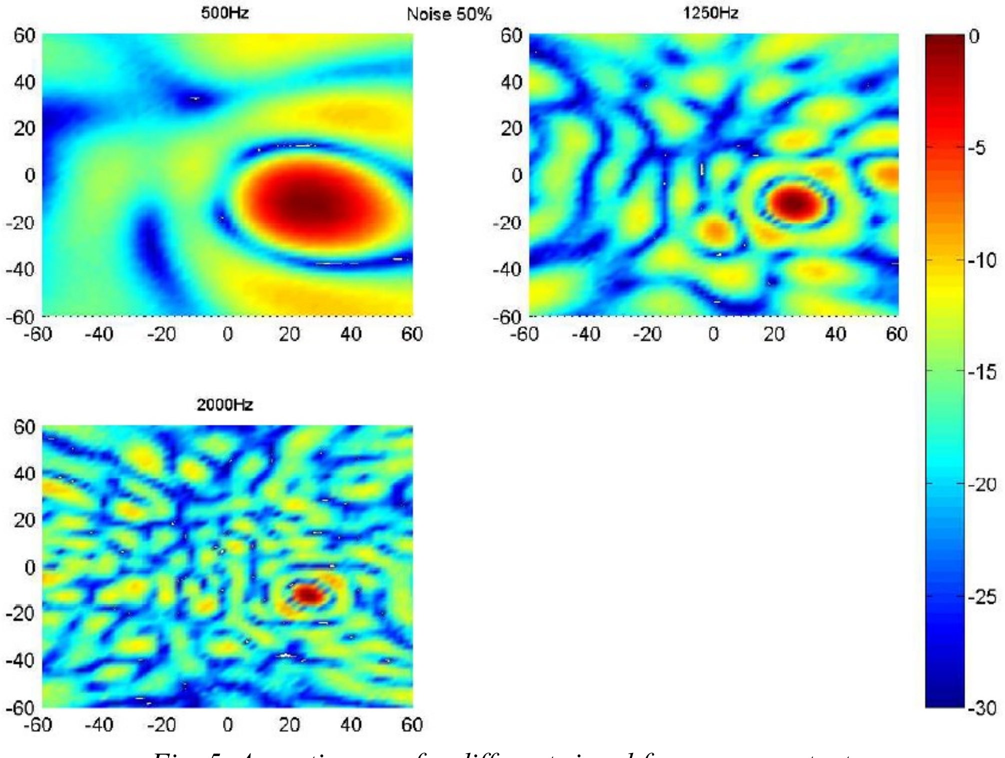


Fig. 5. Acoustic maps for different signal frequency content.

Table 3. FPGA Beamformer main characteristics vs signal frequency content.

	MSL **Maximum Sidelobe Level** [dB]	**Main Lobe Amplitude** [°]
500Hz	-10	26
1250Hz	-8.2	10.3
2000Hz	-8.1	6

5.3 Interference effect

To evaluate the importance of interfering signals, two test scenarios (without and with interference) were evaluated. The source signal in both cases is a sinusoidal signal with a background noise level of 50%. The interfering signal is a signal with the same frequency of the source signal, but coming from a direction with azimuth -10° and elevation of -20°. The results are shown in Fig. 6.

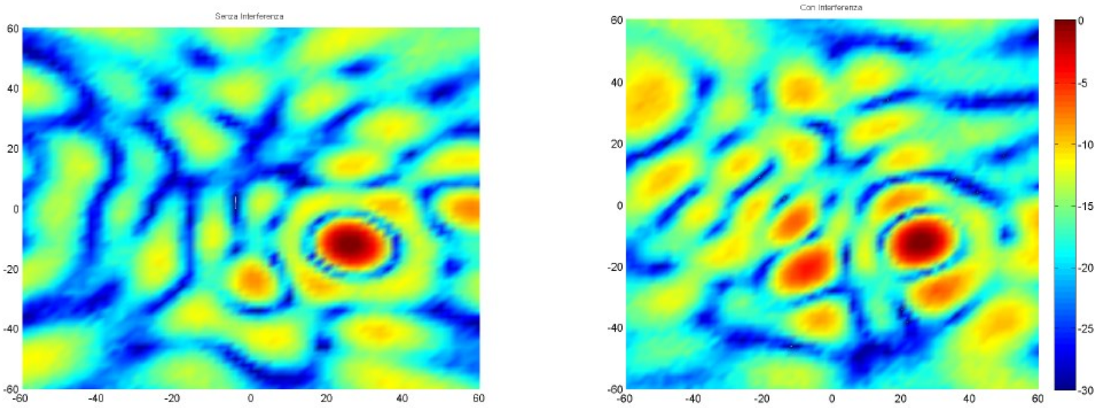


Fig. 6. Acoustic maps for different acoustic scenarios: without interference and a source signal at 1250Hz (left) and the same signal but with an interfering signal at the same frequency and coming from the direction [-10°, -20°].

Table 4. FPGA Beamformer: interfering signal effect.

	MSL **Maximum Sidelobe Level** [dB]	**Main Lobe Amplitude** [°]
1250Hz - No interference	-8.2	10.3
1250Hz - With interference	-7	10.6

It seems that interfering signals have a negligible effect on the main characteristics of the beamformer (i.e. MSL and Main Lobe Amplitude).

5.4 Noise effects

Different background noise levels (from 5% to 100% of the source signal power level) were applied in order to verify the capabilities of the beamformer to filter out the background noise.

The acoustic maps shown in Fig. 7 show that the beamformer is able to filter out incoherent noise arriving to the array sensors. We can also see that adding noise, up to a certain extent, is beneficial to the acoustic maps' characteristics, as it reduces the occurrence of spatial aliasing issues related to the usage of a sinusoidal signal, as we do in our test cases.

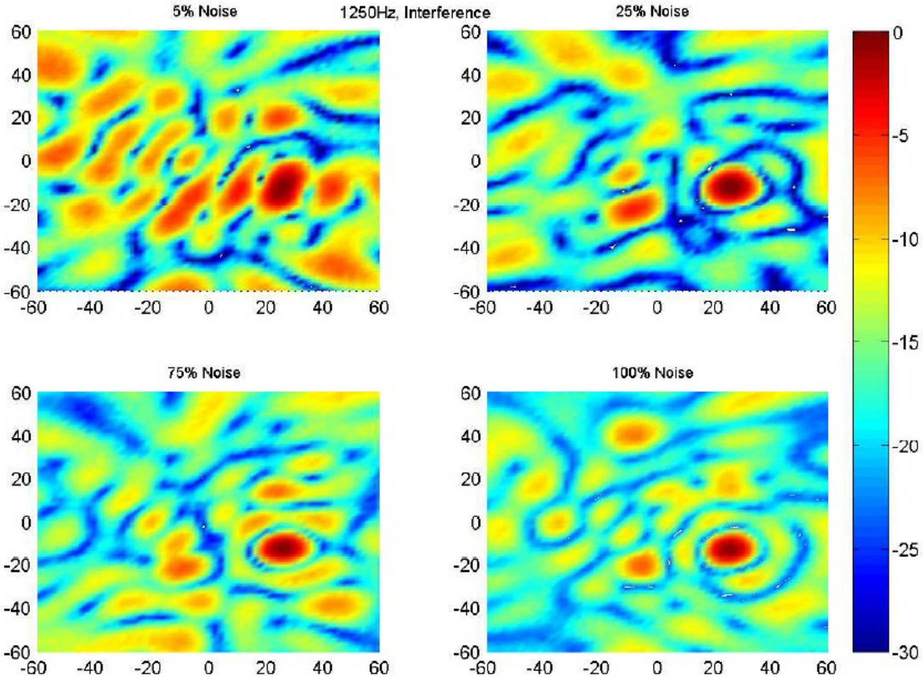


Fig. 7. Acoustic maps for different noise level content.

Table 5. FPGA Beamformer: background noise effect.

Noise Level	**MSL** [dB]	**Main Lobe Amplitude** [°]
5%	-9.0	10.25
25%	-8.9	10.25
50%	-8.2	10.30
75%	-8.7	10.75
100%	-8.7	11.00

5.5 Pointing effects

To see the effect of the steering direction, different tests were performed with the acoustic source positioned precisely on a point of the steering grid and not. The results are shown in Fig. 8. Also in this case, we can see that the effect of non-coincidence of the pointing direction with an acoustic pixel is negligible in the final acoustic map.

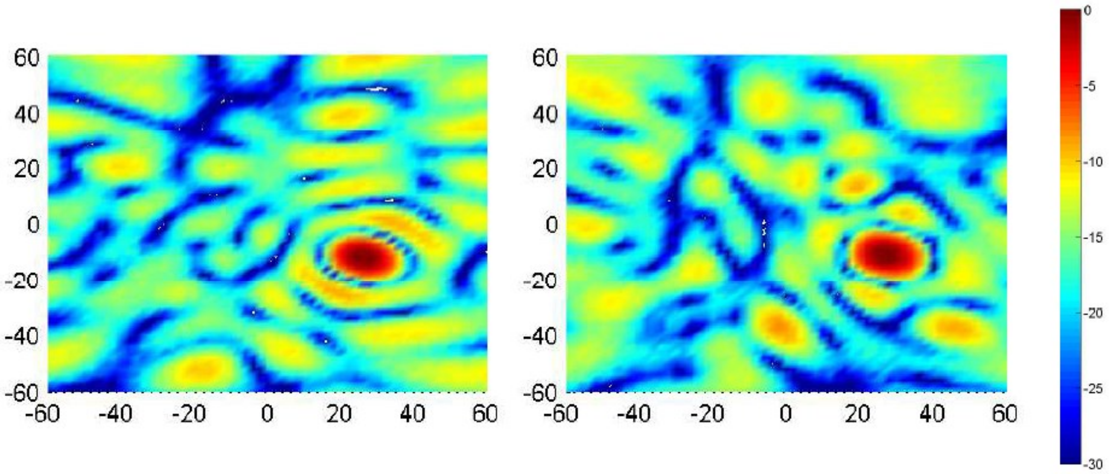


Fig. 8. Acoustic maps for different pointing directions: coincident with a steering grid point (left) and not coincident with a steering grid point (right).

Table 6. FPGA Beamformer: pointing direction effect.

	MSL [dB]	**Main Lobe Amplitude** [°]
1250Hz – Coincident	-8.9	10.25
1250Hz – No coincident	-8.2	10.7

5.6 Broadband Signals

Finally, we present a test performed with an acoustic source representing a real, broadband audio signal. The sample, taken from [9], contains the sound of a car passing by the microphone, at several meters of distance. The signal, in stereo 24-bit Wave format, was converted to mono, and a subsection of 384ms was extracted from it, representing the exact moment in which the car passes by the microphone. The signal was then fed into Matlab's Phased Array Toolbox, simulating a sound source located at the [-20, 20] Az-El coordinates.

The frequency content of the sample is rather wide, with some peaks extending to the 100-150Hz range. As it is widely known, low frequencies are often untreatable in a beamforming context: therefore, we employed here a band-pass filter with cut-off frequencies at 1Khz and 4Khz respectively. The filter's coefficients were then integrated in the constraint matrix C of the LCMV algorithm, so as to be considered during the optimization process.

In Fig. 9 we show a subset of the acoustic maps obtained with the signal described above. It can be seen that the results are very promising, and are again as expected from the theory: the average Main Lobe Amplitude amounts to 9.4° across all acoustic maps, with an average Maximum Side Lobe Level of -12db. These results are better than those obtained with pure tonal signal which, as mentioned earlier, may be subject to spatial aliasing issues.

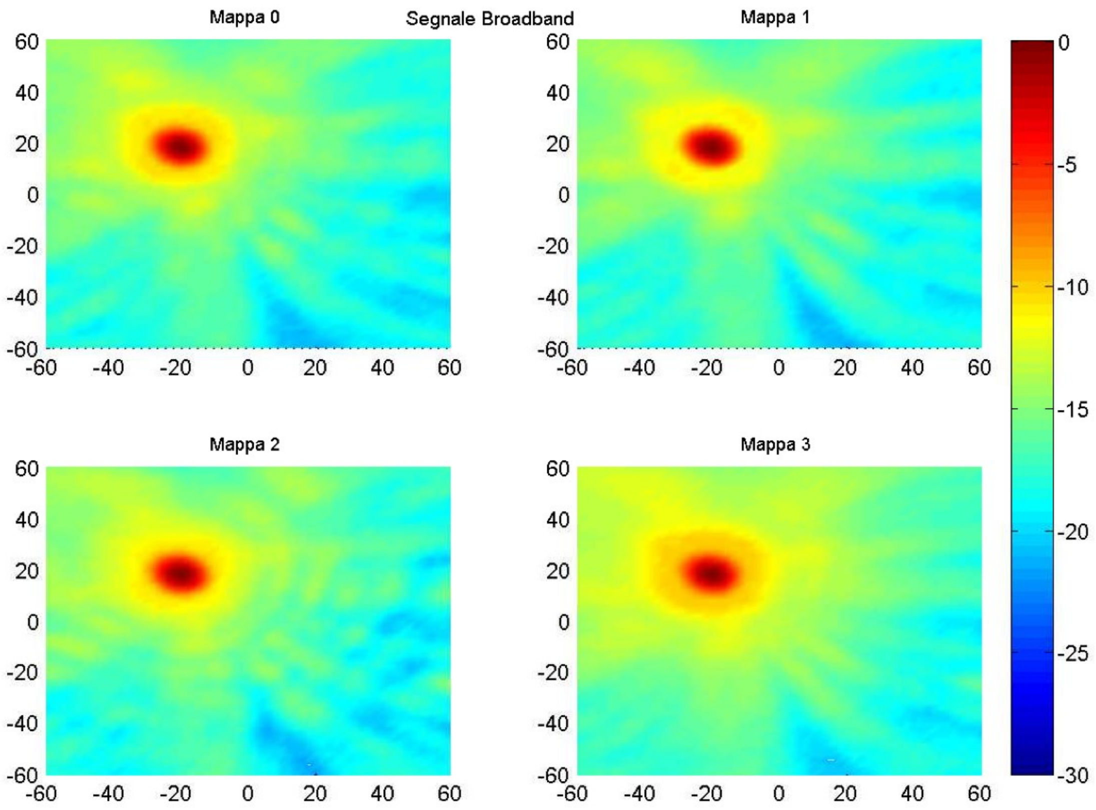


Fig. 9. Acoustic maps obtained from a broadband signal containing pass-by car audio.

6 CONCLUSIONS

The numerical test campaign shown in the present work has demonstrated the following characteristics of our beamformer, as implemented on FPGA architecture:

- the algorithm's accuracy with fixed-point arithmetic (used by FPGA-implemented algorithm) is comparable with the floating point one;
- the algorithm can process both tonal and broadband signals with similar performances;
- interfering noise sources are strongly attenuated;
- high background noise level is correctly filtered;
- spatial localization accuracy is as expected from the theory.

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