UM2214
User manual

Getting started with AcousticBF real-time beam forming middleware

Introduction

The AcousticBF software provides an implementation for a real-time adaptive beamforming algorithm: using the PDM or PCM signals acquired from two digital MEMS microphones, it creates a virtual directional microphone pointing in a fixed direction in space. Several configurations are available, allowing the user to find the best tradeoff between audio output quality and resource consumption. Parameters and modalities can be modified at runtime for immediate adaptation to the varying environmental conditions.

The AcousticBF library is provided in binary format as part of a software package providing sample implementations running on the X-NUCLEO-CCA02M1, when connected to a NUCLEO-F401RE.

The library can be easily ported to any STM32F4 microcontroller with an FPU. The software is based on STM32Cube technology. Information about STM32Cube is available on www.st.com at *http://www.st.com/stm32cube*.

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1 Licensing information

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Some of the library code is based on the CMSIS DSP software library by ARM, a suite of common signal processing functions for use on ARM Cortex-M processor based devices. Licensing terms are available in the release_note.html file included in the software package, in the next lines of this document and on the web at http://www.keil.com/pack/doc/CMSIS/DSP/html/index.html.

Some of the library code is based on the open Source patent-free SPEEX software, by Jean-Marc Valin/Xiph.Org Foundation, released under "revised BSD" license. Licensing terms are available in this document, in the release note for this software, in the header file of this software and on the web at: http://www.xiph.org/licenses/bsd/speex/

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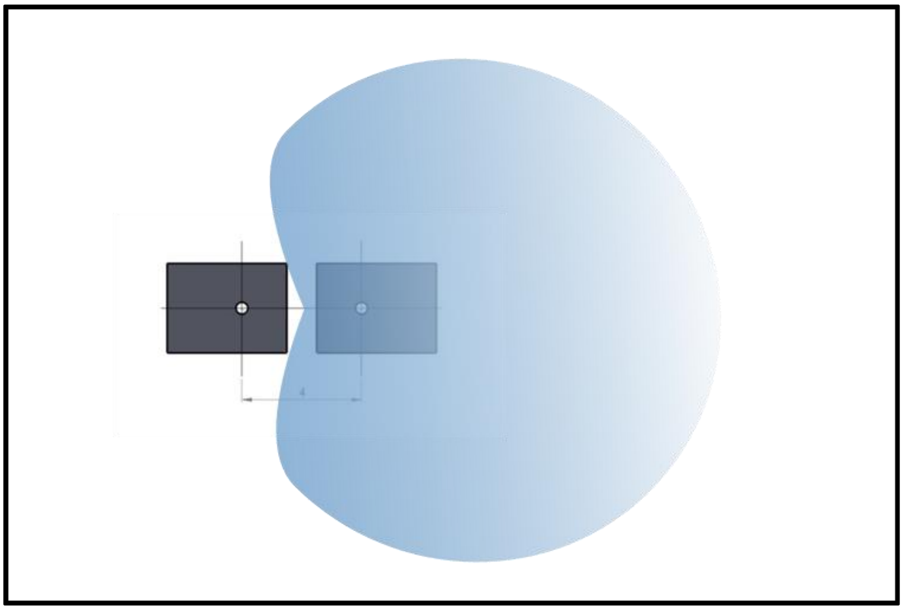
2 AcousticBF software library

This library uses the signals from two omnidirectional MEMS digital microphones like ST MP34DT01-M to create a virtual directional microphone. Several algorithm configurations are available for the user to find the best tradeoff between audio output quality and resource consumption. Parameters and modalities can be modified at runtime for immediate adaptation to the varying ambient conditions.

The audio acquisition system must have:

- two microphones placed at a known distance
- audio data acquired in one of the following formats:
  - standard PDM format
  - PCM format at 16 kHz (in this case, in the current version, microphone distance must be equal to 21 mm)
- STM32 microcontroller with Floating Point Unit (FPU) to acquire the microphones and run the algorithm.

Figure 1: MEMS microphone arrangement and output beam pattern



Microphone distance less than 40 mm gives best results in terms of audio quality and frequency response.

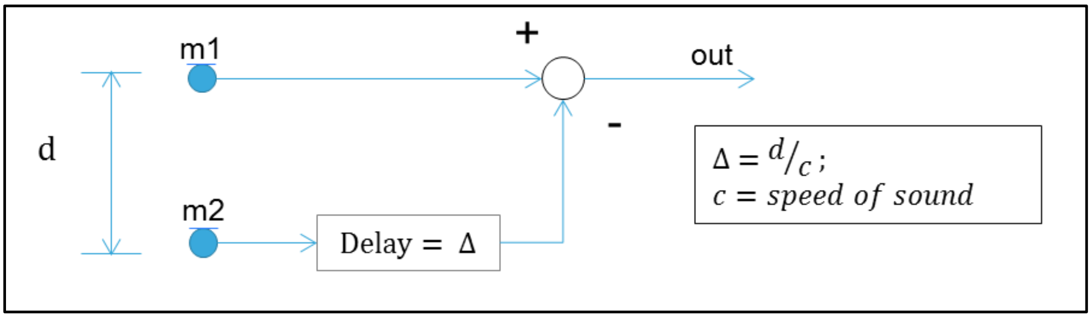
Figure 1: "MEMS microphone arrangement and output beam pattern" shows a typical microphone configuration, where the distance between the acoustic holes of two microphones is 4 mm. The blue area represents a cardioid polar pattern that is symmetrical with respect to a line joining the two microphones (typically referred to as an “endfire” configuration). Inverting the order of the microphone signals input into the library results in a beam oriented in the opposite direction. Different directions can thus be obtained with microphone arrays by feeding the library with different signal couples.

The library takes two PDM or PCM input streams from the respective MEMS microphones and is able to output up to two PCM channels (16 kHz, 16 bits per sample) representing the algorithm output (first channel) and the unprocessed output from one of the two omnidirectional microphones (second channel). The latter may be used as a reference for test and evaluation purposes.

2.1 Description

The library supports a first-order differential microphone array (DMA) based on two omnidirectional digital MEMS microphones, such as MP34DT01. A cardioid beam-pattern is implemented by delaying the sound signal captured by one of the microphones by an amount equal to the acoustic delay between the microphones along the “endfire” direction, as shown in the following picture.

Figure 2: Cardioid differential microphone array



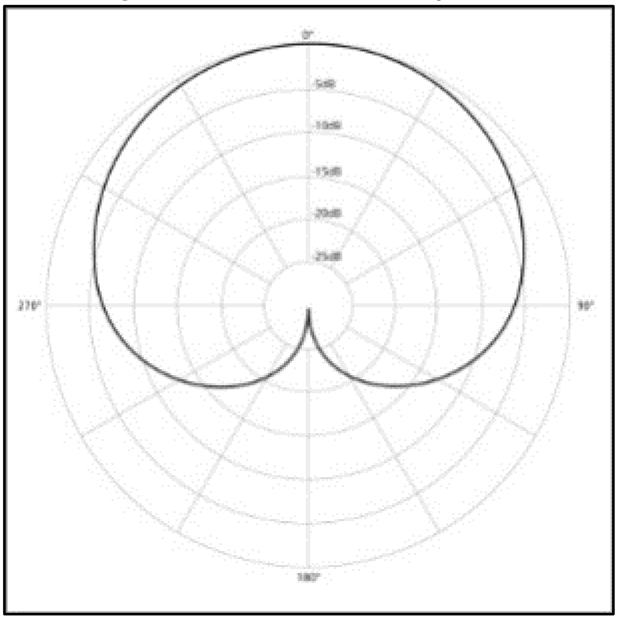
In an extended field context where all the acoustic sources are much further from the microphones than the distance "d" between microphones, the system can be described by the following equation:

Equation 1

$$out(t) = m_1(t) - m_2(t - \Delta);$$

In a typical DMA, the distance between microphones is much shorter than the shortest acoustic wavelength of interest, and the resulting beam pattern is reasonably independent of frequency. *Figure 3: "Ideal cardioid beam pattern"* shows an ideal cardioid beam pattern.

Figure 3: Ideal cardioid beam pattern



2.2 Processing options

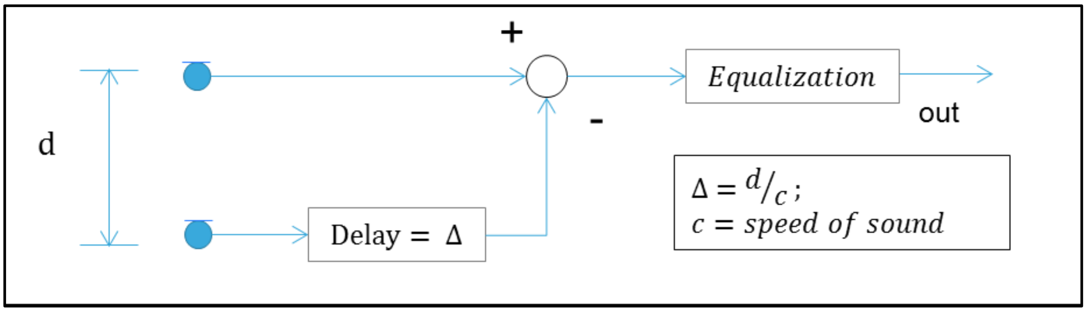
The beamforming function implements a modular composition of functional blocks. Its overall complexity is scaled into four levels of algorithm intensity; each of them implementing a function subset. The library can therefore match different user requirements, by allowing levels of trade-off between resource consumption and output quality (both in terms of directivity and SNR).

2.2.1 Cardioid basic

When the library is initialized using this option, a cardioid beam former is implemented based on a first-order differential microphone array (DMA).

The DMA configuration alters the frequency response of individual microphones, introducing high-pass behavior rising in frequency at 6 dB/octave. For this reason, a filtering stage is added to flatten the system response in the audio frequency range.

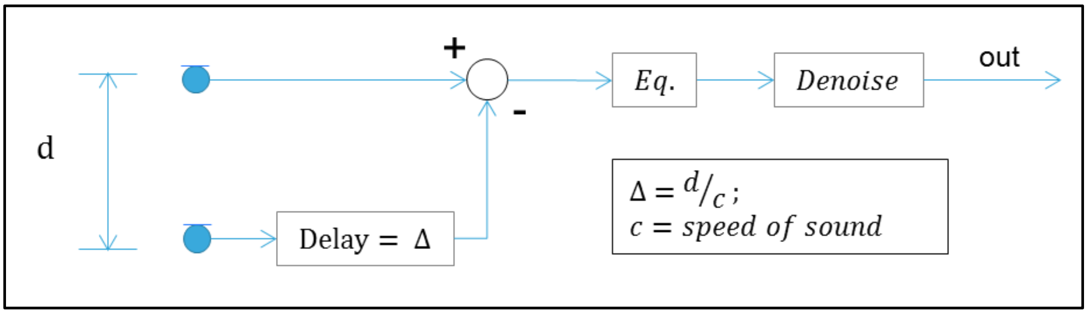
Figure 4: Cardioid basic block diagram



2.2.2 Cardioid denoise

The basic cardioid setup above produces audible noise, particularly at low frequencies, which could be undesirable for certain applications. For this reason, the Cardioid denoise option activates a de-noising filter applied to the output of the cardioid beamforming.

Figure 5: Cardioid denoise block diagram

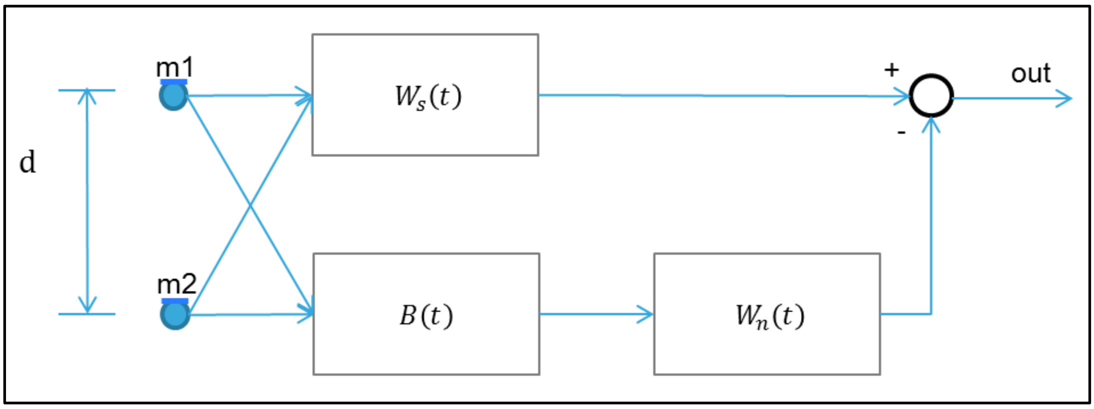


2.2.3 Strong

The “Strong” option activates a generalized sidelobe canceller (GSC) algorithm adapted to the differential microphone array architecture. With this approach, the output of a cardioid beamformer aimed at the desired source is merged with another signal, defined by a ‘blocking matrix’, which is adaptively filtered so as to minimize the power of unwanted audio components when the two signal paths are recombined[a].

[a] McCowan, I. (2001). Microphone arrays : A tutorial.

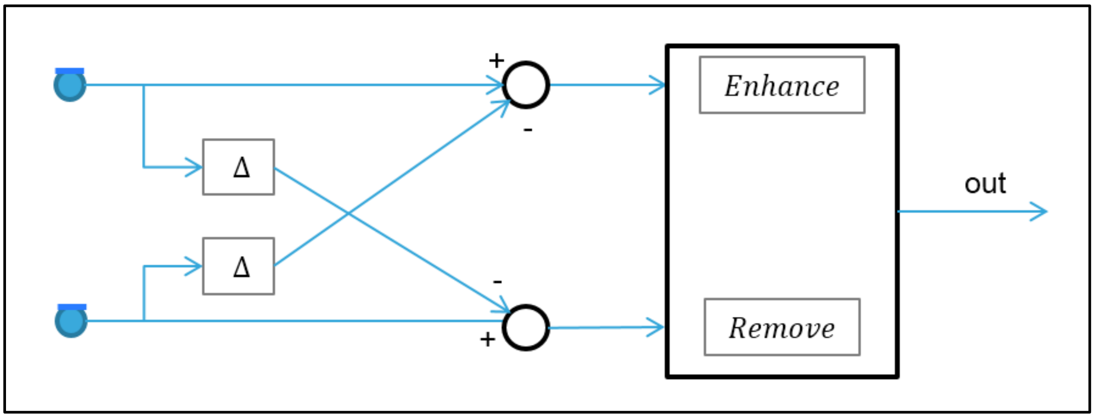
Figure 6: Generalized sidelobe canceller - block diagram



In the figure above, the $W_s(t)$ operation implements the cardioid processing scheme shown in *Figure 7: "Optimization: Strong - block diagram"*, and operator implements a symmetrical schema where the microphones are exchanged so as to aim the beam in the opposite direction, with a 'zero' of the beam pattern towards the actual source. This configuration is known as back to back cardioid[a].

In the implementation of the "Strong" type, an adaptive filter fed with back to back cardioid signals removes the noise components from the target source and outputs a signal characterized by a very directive pattern. The last processing step is the application of a de-noising filter, like in the Cardioid denoise option, which removes residual noise.

Figure 7: Optimization: Strong - block diagram

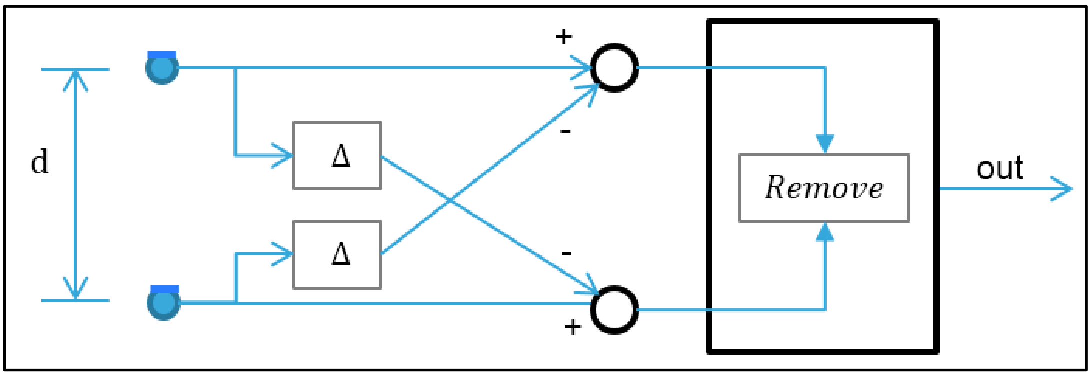


2.2.4 ASR ready

The ASR ready option is a subset of the strong optimization, in which the final de-noising filter is not activated. This produces an output signal characterized by the same directivity pattern as "Strong", but residual noise is not removed. This leads to savings in terms of resources without affecting the ASR performance. For ASR performance tests, please refer to *Section 2.4.2: "ASR test"*.

[a] Mingsian R. Bai, J.-G. I. (2010). Acoustic Array Systems: Theory, Implementation, and Application. Wiley.

Figure 8: Optimization: ASR - block diagram



2.3 Microphone matching

To achieve the best results, the microphone sensitivities must be properly matched. The beamforming library allows the setting of a calibration parameter representing the gain difference between the two microphones.

2.4 Tests

Polar pattern and ASR tests regarding the beamforming polar pattern and overall output quality were performed in an anechoic chamber.

2.4.1 Polar pattern

To evaluate the shape of the beam pattern created by the algorithm, a testing environment was set up in an anechoic chamber using a high quality loudspeaker and a rotating support with a 4 mm differential microphone array mounted on top. The microphone array was rotated manually in steps of 10 degrees, and Gaussian white noise was played by the loudspeaker. For comparison purposes, the system was set up so to generate three algorithm outputs at the same time:

1. the omnidirectional signal of one of the two microphones composing the subsystem
2. one of the two cardioids created (no optimization)
3. the overall system (ASR-ready optimization)

The "cardioid denoise" and "strong" enhancements were not used in this test because their output polar pattern is exactly the same as in the"basic cardioid" and "ASR-ready" optimizations, respectively (the final denoising steps don't affect the polar pattern). The power of the acquired audio signal was computed on a host for each direction of arrival and for each array output. The figure below shows the respective polar patterns for omnidirectional microphone (blue), Basic cardioid (red), and ASR-ready (green).

Figure 9: Beam patterns

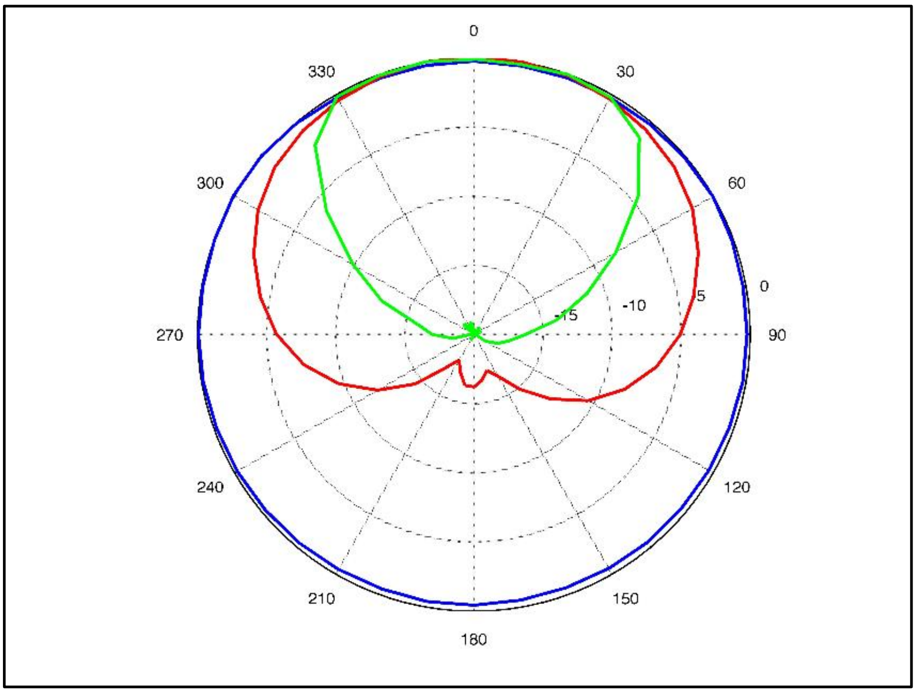
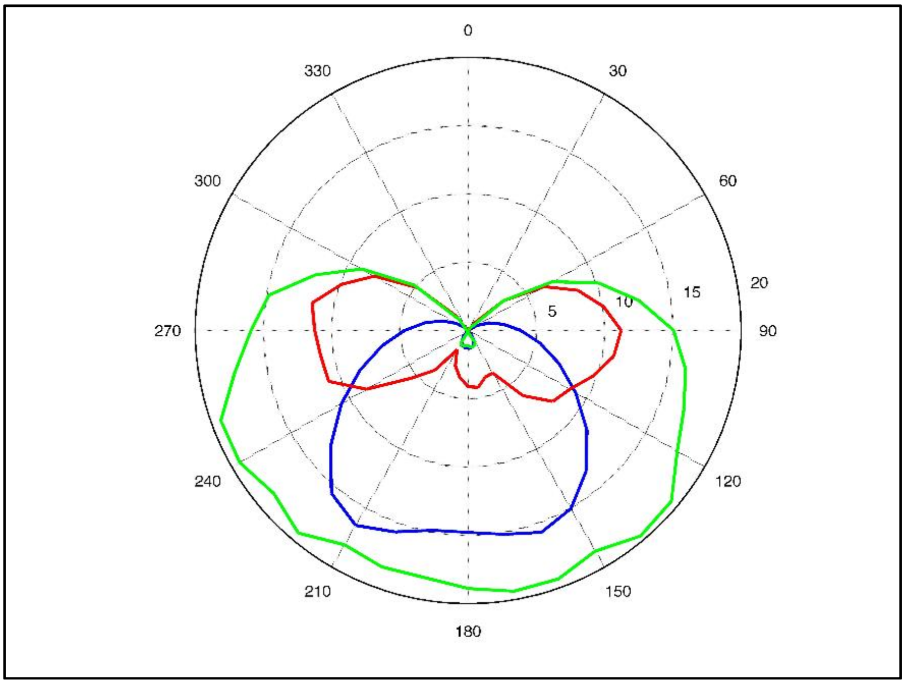


Figure 10: "Improvement in directionality" shows a comparative analysis of the improvement in directionality gained with each type of processing:

- The green line represents the directionality improvement of the overall system (ASR-ready optimization) over the single omnidirectional microphone.
- The red line shows the improvement of the ASR-ready optimization over the standard DMA cardioid beamforming (no optimization).
- The blue line represents the comparison between standard cardioid beamforming (no optimization) and single omnidirectional microphone.

Figure 10: Improvement in directionality

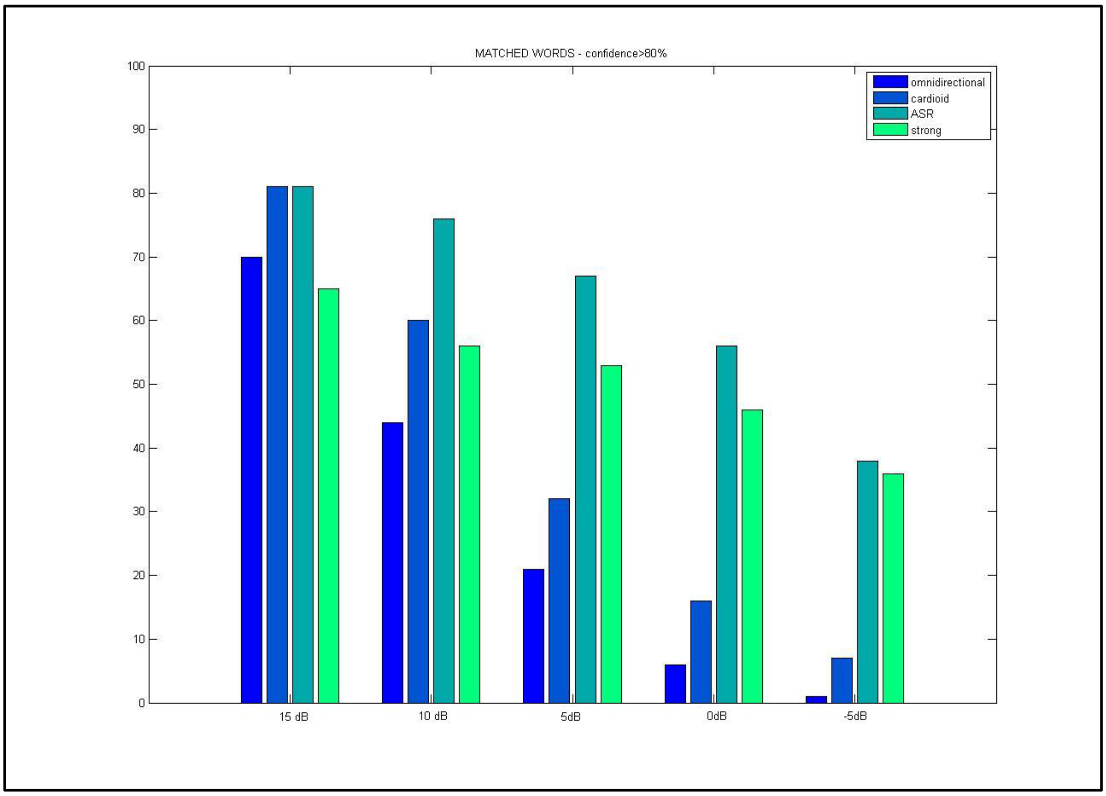


2.4.2 ASR test

For this test, a series of isolated words were reproduced by a high quality loudspeaker placed in the beam direction (0 degrees in the polar pattern), while white noise was reproduced at 90 degrees. Several measurements were taken with changes to the ratio between the noise and voice and using different data sets of words chosen from the same phonetic groups in order to minimize the difference between word intelligibility. The recorded signals were sent to the online Google ASR service for evaluation.

The following figure shows how the system improves ASR performance even in low signal-to-noise ratio (SNR) scenarios.

Figure 11: ASR test results



3 Library profiling

Profiling was performed in order to evaluate the library resource consumption in terms of MIPS, RAM and Flash. Detailed information can be found in the AcousticBF_Package.chm compiled HTML file in the Documentation folder.

4 References

1. McCowan, I. (2001). Microphone arrays : A tutorial.
2. Mingsian R. Bai, J.-G. I. (2010). Acoustic Array Systems: Theory, Implementation, and Application. Wiley.
3. User manual, UM1900 - Getting started with the digital MEMS microphone expansion board based on MP34DT01-M for STM32 Nucleo. Freely available on www.st.com.

5 Revision history

Table 1: Document revision history

Date	Version	Changes
18-May-2017	1	Initial release.