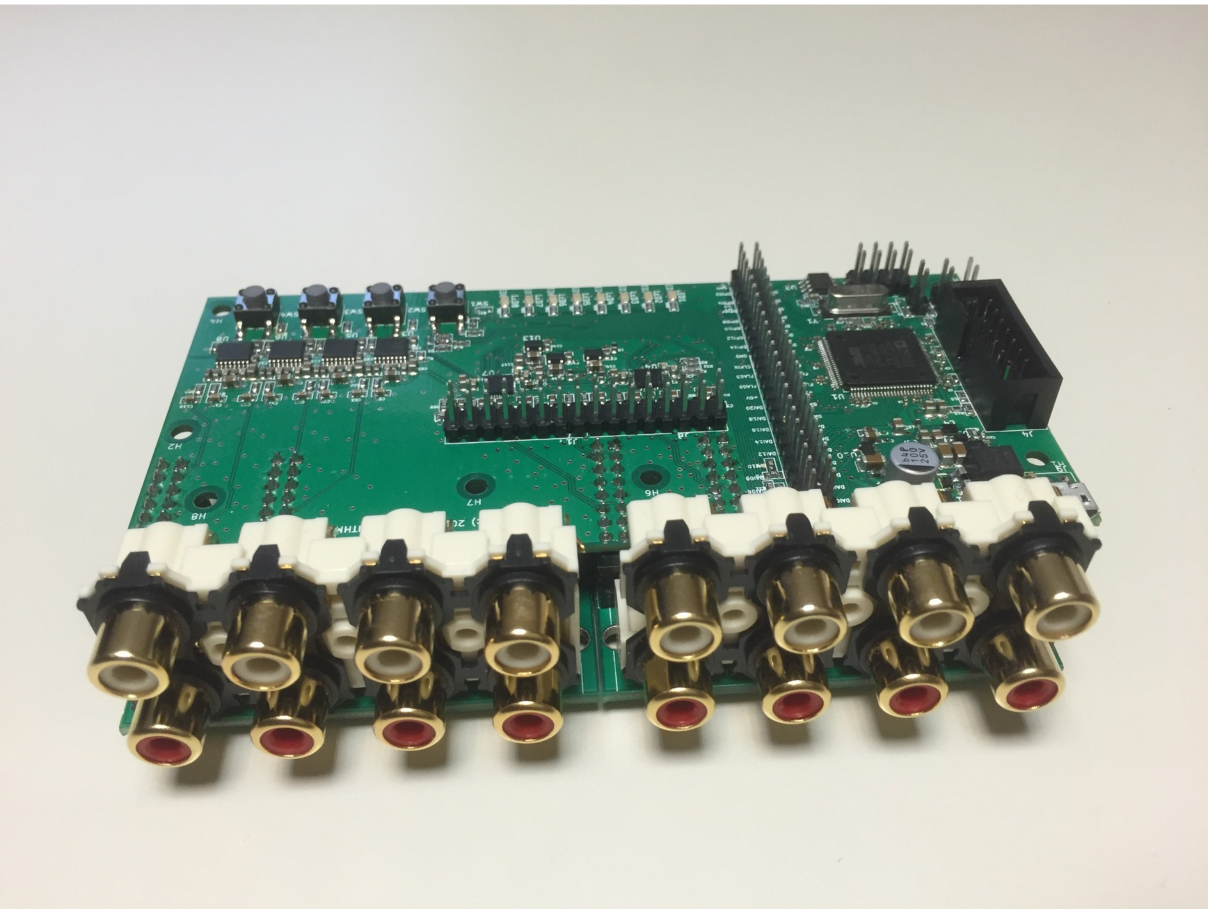

OCEAN-ADSP21489-0808HC
Noise Management

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1 Introduction

This document describes the software features of the noise management solution on the OCEAN-ADSP21489-0808HC audio real-time processor. The hardware is based on the Analog Devices ADSP-21489 SHARC floating-point Digital Signal Processor (DSP). The software is based on the Noise Management library; providing advanced adaptive noise reduction algorithms. Both hardware and software are developed, maintained, and marketed by DSP Algorithms.

The OCEAN-ADSP21489-0808HC hardware includes all the necessary components that efficiently implement a complete stand-alone multi-channel audio application. The hardware includes an Analog Devices ADSP-21489 SHARC processor, 8 analog input channels, 8 analog output channels, and simple user interface consisting in 4 buttons and 8 Light Emitting Diodes (LED). Full control (through for instance a web browser) is also possible using an external microcontroller which communicate with the SHARC DSP through SPI or UART. The hardware is described in a separate document and can be downloaded from https://www.dspalgorithms.com.

The application programs are stored in the on-board non-volatile flash memory. On powering the hardware, the processor copies the software from non-volatile memory to internal RAM and starts executing the program instructions. The real-time application reads the 8 analog audio inputs, processes the audio samples through the audio algorithms, and plays the processed audio samples to the 8 analog outputs.

The analog inputs and outputs on the OCEAN-ADSP21489-0808HC are provided through industry standard 0.1" headers. Those headers can be used to offer the inputs and outputs on a variety of audio connectors, such as XLR, RCA, jacks, or terminal blocks, as needed by the end user requirements to interface to the microphones and loudspeakers.

2 Noise Management Software applications

The OCEAN-ADSP21489-0808HC hardware is designed with high-end audio applications in mind. Therefore, several noise management applications that can be directly used in field testing are available. The current release of the OCEAN-ADSP21489-0808HC Noise Management firmware stored in the flash memory includes two applications.

1. 6-Channel Microphone Array: This application boots by default when the OCEAN-ADSP21489-0808HC is powered on and expects 6 microphones (for instance both OCEAN-03-MIC-LEFT and OCEAN-03-MIC-RIGHT) to be properly connected to input channels 3 to 8. The microphone signals are processed through the Adaptive Interference Cancellation, BRIL noise reduction, and then combined using the beamforming algorithm. This application is described in details in Section 2.3.

2. 6-Channel Line-in Array: To boot this application press `SW1` while powering the OCEAN-ADSP21489-0808HC hardware. You must disconnect the linear array microphones and connect 6 external microphones with their external amplifiers to channels 3 to 8 before booting this application. The microphone signals are processed through the Adaptive Interference Cancellation, BRIL noise reduction, and then combined using the beamforming algorithm. This application is described in details in Section 2.4.

All Noise Management applications run at 48 kHz sampling rate. Audio data processing is performed on blocks of 128 samples (2.67 ms blocks at 48 kHz sampling rate). While one block of data is collected in an input buffer, the previously collected input buffer is processed, and the already processed buffer is sent to the output. This scheme of block processing introduces buffering delay of twice the block length, or 2x128=256 samples (5.33 ms).

All noise management applications are similar in structure and differ only in the input and output connections and whether the microphone bias and gain are provided by the OCEAN-ADSP21489-0808HC hardware or by external hardware. Each application is divided into four different sections, namely Initialization, Audio data input and output, Data processing, and User Interface.

Initialization: The initialization stage is responsible for setting up the hardware, preparing the ADC and DAC for audio data conversion, and setting up the input signal conditioning as required by the specific application.

Audio data input and output: The audio signals at the 8 analog inputs are sampled by the Analog to Digital Converters and transferred to the processor's internal memory during an Interrupt Service Routine (ISR) using Direct Memory Access (DMA). In the same (ISR), the processed audio data is transferred from the processor's internal memory to the 8 Digital to Analog Converters. When a complete new buffer of 128 samples arrives, this buffer is processed, the previously processed buffer is sent to the output sample per sample, and new input samples are collected in an empty buffer.

Data Processing: This is where the application calls audio processing functions to reduce noise and improve signal to noise ratio. All processing must be completed within the time required to collect a new buffer (2.67 ms at 48 kHz sampling rate) to meet the hard real-time requirement.

User Interface: The software includes several processing modules, each of which may be enabled or disabled at run-time by calling a simple API. The OCEAN-ADSP21489-0808HC push-buttons are used to implement this functionality. At the same time, the OCEAN-ADSP21489-0808HC LEDs are used to indicate when the input signals are overloaded (level exceeds ADC maximum allowed input level) and when a function is enabled (LED on) or disabled (LED off).

Although the LEDs and buttons provide basic user interface, gaining full control can be achieved only using an external microcontroller. All noise management applications support SPI commands by default and fully functioning web interface is available to enable/disable each and every processing function, set/get each algorithm parameter, display performance statistics, and plot adaptive filter coefficients in real-time. This web user interface is described in a separate document.

The following sections describe the noise management algorithms included in the firmware in more details.

2.1 Noise Management Library

DSP Algorithms Noise Management library is a mature software-only speech enhancement library. The library includes all digital signal processing algorithms necessary to create high quality audio applications. The Noise Management library includes the following processing modules (see also Figure 1).

Both the noise reference signals as well as the microphones signals are processed with different processing modules. Each processing module can be enabled, disabled, and its target level (when applicable) can be changed at any time during real-time operation. Enabling, disabling, and setting the target level takes effect immediately without any delay.

The Noise Reference signals are needed for the Adaptive Interference Cancellation algorithm and pick up signals correlated to the noise need to be canceled. The Noise Reference signals can be generated using microphones placed far from the users to avoid picking up users' voices. Alternatively, they can be generated using another type of transducer that pick up a noise reference but no user speech such as an accelerometer, vibration sensor, or tachometer. The Noise Reference signals can be conditioned using the following processing modules.

- DC blocking filter (DCF) which removes any constant bias DC voltage from the sensor signal.
- Graphic Equalizer (GEQ) which is a set of band-pass filters each with its own adjustable gain and can be used for instance to correct the frequency response of the sensor to improve the adaptive interference cancellation performance.

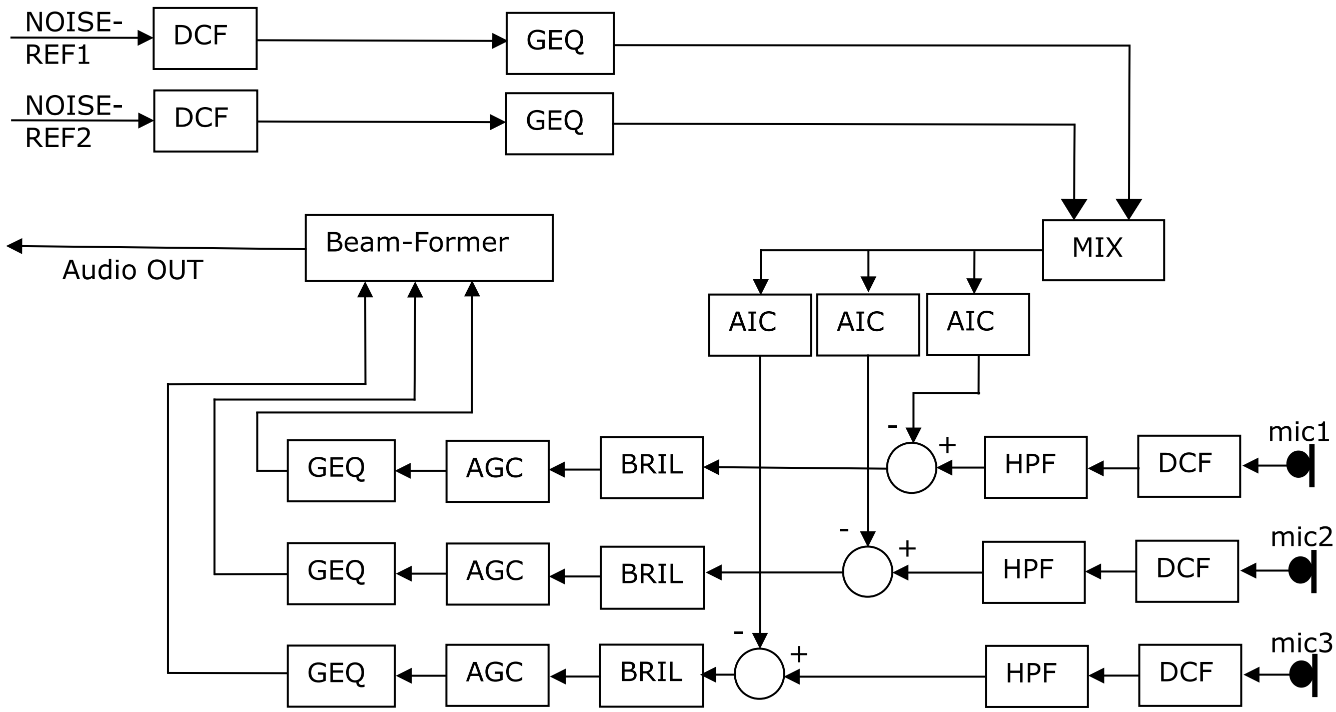

Figure 1: Block diagram of the Noise Management processing modules.

The signals picked up by the microphones which contain the local users' voices as well as noise, and reverberation are processed through the following processing modules.

- DC blocking filter (DCF) which removes any constant bias DC voltage from the microphone signal.
- High Pass Filter (HPF) which removes the very low frequencies from the microphone signal. The contents below the HPF corner frequency usually include noise and hardly any speech information.
- Adaptive Interference Canceller (AIC) which estimates the noise picked up by the microphone (based on the noise references) and subtracts this estimate from the microphone signal to remove as much of the noise as possible without distorting the local user's speech. The AIC module is implemented using a fast converging, robust, and efficient adaptive filter algorithm. The AIC is well protected from divergence during the users speech periods, therefore provides very high quality processed signal.
- The BRIL algorithm is a stationary noise reduction algorithm and is used to reduce the residual noise left after each AIC channel. Unlike AIC, which introduces no distortion whatsoever to the microphone signal, BRIL may introduce some distortion, and therefore properly setting the BRIL noise reduction level is important.
- Automatic Gain Controller (AGC) which keeps each microphone signal at a constant level that is user-adjustable. Since the microphone signal can vary significantly in level depends on the microphone sensitivity and the distance from the user, the AGC can be set to amplify or attenuate the incoming signal to a pre-defined constant level.

- Graphic Equalizer (GEQ) which is a set of band-pass filters each with its own adjustable gain and can be used for instance to correct the frequency response of the microphone, the grill in front of the microphone, or simply to adjust certain frequency bands of the speech to pre-specified characteristics.
- The beamformer combines all microphone signals in an optimum way to enhance the local users speech while reducing the noise and reverberation. The BF can be designed to be fixed or fully adaptive, as the application needs. It works by filtering each microphone signal through a filter and all filtered signals are then summed together. The filters are designed to achieve optimum sound quality from all microphones combined.

2.2 Noise Management User Interface

All noise management applications on the OCEAN-ADSP21489-0808HC have the same user interface that makes use of the on-board LEDs and push-buttons. It is important to mention that the user interface described below is designed to allow a stand-alone application. An OEM application based on this OCEAN-ADSP21489-0808HC may, for example, use a microcontroller to send commands to the DSP over the SPI header to achieve full control user interface. This latter option (web interface through SPI or UART) is described in a separate document. This section describes only user interface implemented using the OCEAN-ADSP21489-0808HC LEDs and buttons.

The noise management applications use the four push-buttons and eight LEDs to implement a few capabilities of the Application Programming Interface (API) of the underlying software library. The function of each button and LED are described below.

`SW1` button is used to start and stop the audio real-time processing. The application always starts in STAND-BY mode. In this mode the software just blinks `LED1` and `LED2` simultaneously at the rate of 0.25 Hz, no real-time processing of audio signals is performed. After booting, the application switches from STAND-BY to audio processing automatically after 3 seconds. AT any time pressing `SW1` while in STAND-By will reset the DSP and start the real-time operation. During real-time operation, `LED1` and `LED2` switch ON and OFF alternatively at the rate of 1 Hz.

`SW2` button is used to increase BRIL noise reduction level by 1 dB (more noise reduction) on all microphones. The applications starts with the Noise Reduction Level at 10 dB by default. Each Press on `SW2` will increase the Noise Reduction Level by 1 dB and `LED4` will turn ON briefly as visual confirmation of the increase.

`SW3` button is used to decrease BRIL noise reduction level by 1 dB (less noise reduction) on all microphones. The applications starts with the Noise Reduction Level at 10 dB by default. Each Press on `SW3` will decrease the Noise Reduction Level by 1 dB and `LED5` will turn ON briefly as visual confirmation of the decrease.

`SW4` button is used to Enable or Disable the Adaptive Interference Canceller (AIC) on all microphones. The applications start with the AIC disabled by default. If signal connections and levels are correct, you should hear the noise rapidly reducing after pressing `SW4`. When AIC is enabled, `LED3` will turn ON. Pushing `SW4` button when `LED3` is ON will disable AIC, `LED3` will turn OFF, and you should hear sudden increase in noise on all outputs. Pushing `SW4` button while LED3 is OFF enables AIC, LED3 will turn ON, and you should hear noise decrease on the outputs.

The applications also check the level of the input signals before processing each block of audio samples. If one or more input signal exceed the analog to digital converter maximum analog input level, a visual indication is given by temporarily turning ON one of the LEDs. LED7 is used to indicate that one or more microphone signal is too high. LED8 is used to indicate that one or more of the noise reference signals (AIC reference) is too high.

The table below summarizes the Noise Managemnt applications user interface.

push-button	Function	Visual Indicators
SW1	Toggles between STAND-BY and RUN modes	LED1,LED2
SW2	Increases Noise reduction level by 1 dB	LED4
SW3	Decreases Noise reduction level by 1 dB	LED5
SW4	Toggles Adaptive Interference Cancellation ON and OFF	LED3
	At least one microphone signal is too high	LED7
	At least one noise reference is too high	LED8

Remarks:

- The Adaptive Interference canceller will not work properly if either the noise reference or microphone signals levels are too high. This is because the signals are then saturated and clipped when converted to digital form causing severe non-linearity. Reduce the levels until both LED7 and LED8 turn OFF completely, reset the application by pressing SW1 **twice**, and try again.

- It is essential to reset the application before each use to ensure that the algorithms starts off from a well known initial state. You can reset by pressing SW1 to enter the STAND-BY state then pressing SW1 again to reset and start real-time processing. Alternatively, you may also power cycle the OCEAN-ADSP21489-0808HC.

- SW2, SW3 buttons increase and decrease noise reduction level in the range 0 to 40 dB. Pressing SW2 while noise reduction level is at 40 dB has no effect. Similarly, pressing SW3 while noise reduction is at 0 dB has no effect.

- The beamformer will not converge while there are continuous signals from the Noise Reference sensors. To correctly use the beamformer, disconnect the noise reference and start talking for a few seconds, then reconnect the noise references again for normal operation.

- Adaptive Interference Canceller reference signals are set to receive line level signals. This allows using any sensor that can generate signals correlated with the noises that need to be reduced. If microphones (or other sensors that generate week signals) are used to generate the AIC reference signals, those weak signals must be amplified to Line level using a good quality (low noise) external preamplifier.

- For the Adaptive Interference Canceller to work properly, the noise reference must be correlated only with the noises that need to be reduced. If the noise reference is also correlated with user speech, speech distortion may occur.

2.3 6-Channel Microphone Array (default)

This application boots by default as soon as the OCEAN-ADSP21489-0808HC is powered on and requires the OCEAN-ADSP21489-0808HC main PCB and the 6-microphone linear array (OCEAN-03-MIC-LEFT/RIGHT) PCB as well. The microphones PCBs plug directly into the OCEAN-ADSP21489-0808HC Analog Input Header IN3 to IN8, where the microphones also receive proper low noise biasing.

This application can process up to two Adaptive Interference Cancellation reference signals connected to the OCEAN-ADSP21489-0808HC at inputs IN1 and IN2 (line level). The local speech is collected using 6 microphones arranged on one line separated by 5cm apart on the microphones PCB. The microphone signals are fed to the OCEAN-ADSP21489-0808HC at inputs IN3 to IN8 (so no connections on the audio input connectors IN3 to IN8 are required). The following table and Figure 2 summarize the input connections.

Input	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8
signal	REF1	REF2	MIC1	MIC2	MIC3	MIC4	MIC5	MIC6

At the output side, each of the individual processed microphone channel is sent to OUT1 to OUT6. The beamformer output is sent to OUT8. The following table and Figure 2 summarize the output connections.

Output	OUT1	OUT2	OUT3	OUT4	OUT5	OUT6	OUT7	OUT8
signal	MIC1	MIC2	MIC3	MIC4	MIC5	MIC6	NOT USED	BF-OUT

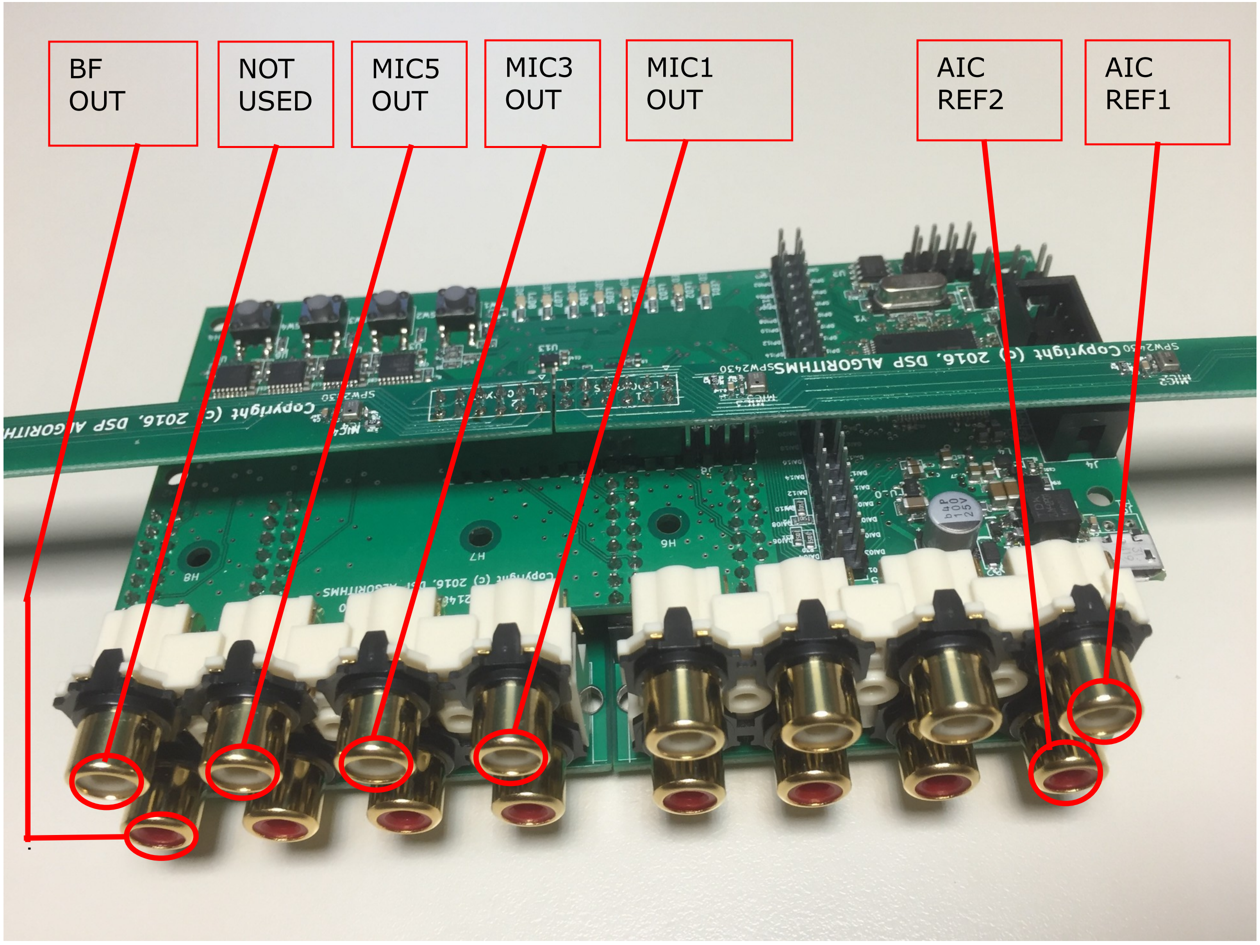

Figure 2: Connections for 6-Channel Mic Input Application.

2.4 6-Channel Line-in Array

In some cases, you may need to use special microphones with their external amplifiers to verify certain testing conditions. In those cases it is preferable to connect those microphones to the OCEAN-ADSP21489-0808HC instead of the linear array MEMS microphones. The 6-Channel Line-in Array application is set up to allow such tests. To boot the 6-channel line-in Array application, hold down `SW1` while powering the OCEAN-ADSP21489-0808HC hardware. This application is identical to the 6-Channel Microphone Array except that it accepts only Line level signals at all inputs.

This application requires the main OCEAN-ADSP21489-0808HC PCB only, the microphones PCB is not needed and should be disconnected from the Analog Input Header. In this use scenario the two AIC reference signals are fed to the OCEAN-ADSP21489-0808HC at input IN1 and IN2. The local speech signals are collected using 6 external microphones positioned anywhere in the listening space. The microphone signals are amplified using external microphone amplifiers and fed to the OCEAN-ADSP21489-0808HC at inputs IN3 to IN8. Inputs IN3 to IN8 are configured as line level inputs in this application (internal PGA gain = 0 dB). The following table and Figure 3 summarize the input connections.

Input signal	IN1	IN2	IN3	IN4	IN5	IN6	IN7	IN8
	REF1	REF2	MIC1	MIC2	MIC3	MIC4	MIC5	MIC6

At the output side, each of the individual processed microphone channel is sent to OUT1 to OUT6. The beamformer output is sent to OUT8. The following table and Figure 2 summarize the output connections.

Output signal	OUT1	OUT2	OUT3	OUT4	OUT5	OUT6	OUT7	OUT8
	MIC1	MIC2	MIC3	MIC4	MIC5	MIC6	NOT USED	BF-OUT

When using the audio connectors to connect the microphones to the OCEAN-ADSP21489-0808HC, the -IN of input channels 3 to 8 must be connected to the analog ground. This can be achieved by placing a jumper between each channel -IN and ground pins on J8 and J14, as shown in Figure 3.

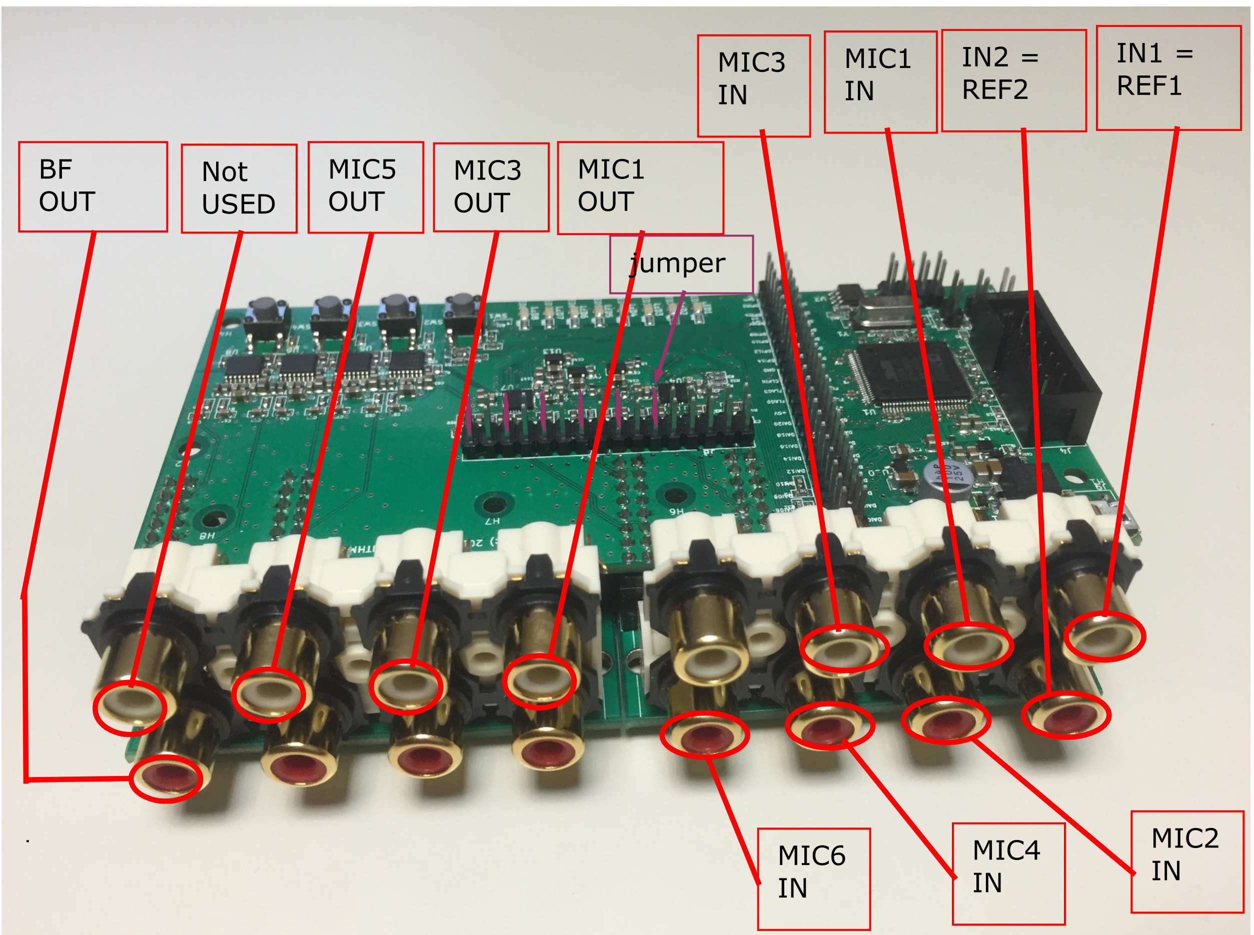

Figure 3: Connections for the 6-channel Line Input application.