

# Acoustic Beamforming for Sound Recording in Noisy Classroom Environments

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**Problem: Recording in Noisy Classrooms**

The complex and dynamic nature of classrooms often prevents effective and non-obtrusive observation and audiovisual recording of authentic learning activities. Up-close positioning of recording equipment may alter student behavior and can increase the complexity of post-processing and data analysis.

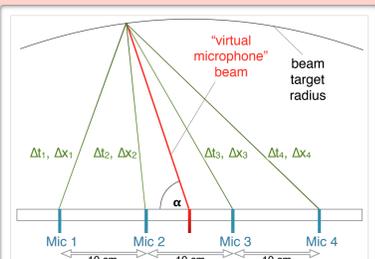


**Possible Solution: Acoustic Beamforming**

Acoustic beamforming enables unobtrusive recording and comprehensive observation through amplification of desired signal and attenuation of unwanted noise.

**Linear Microphone Array**

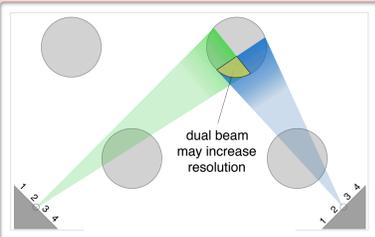
Allows for superposition of time-delayed signals from same signal source. "Spatial filtering" allows separation of desired from unwanted signals.



$\alpha$ : angle of beam with respect to array  
 $\Delta x_i$ : Distance between signal source and respective microphones  
 $\Delta t_i$ : Time elapsed between signal generation and signal reception

**Classroom Audio Recording**

Array positioned at perimeter for unobtrusive recording. Increasing number of microphones can improve spatial resolution. Using multiple arrays may further increase flexibility and resolution of recording to narrow in on individual speakers. **Initial tests were conducted with a single array only.**



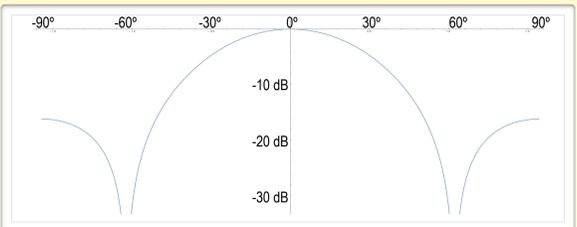
**Realignment of audio tracks from a multi-track recording creates selectively constructive interference to amplify signals from specific portions of the recording space and attenuate others to separate desired from unwanted signals.**

**Equipment: Microphone Array Prototype**

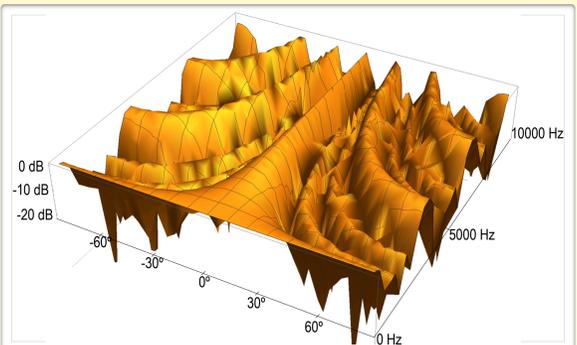
Zoom F8 multi-track audio recorder and four Audio-Technica Pro 45 microphones mounted in linear array.



**Microphone Array Characteristics**



Theoretical sensitivity of the array shown for 1 kHz, 90° (default) beam angle. 10 dB attenuation corresponds to half loudness.

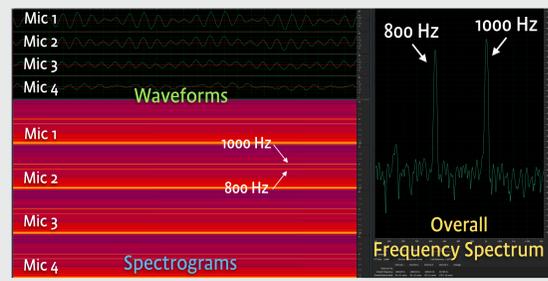


Array sensitivity is frequency-dependent. Therefore, spatial filtering ability depends on angle **and** frequency.

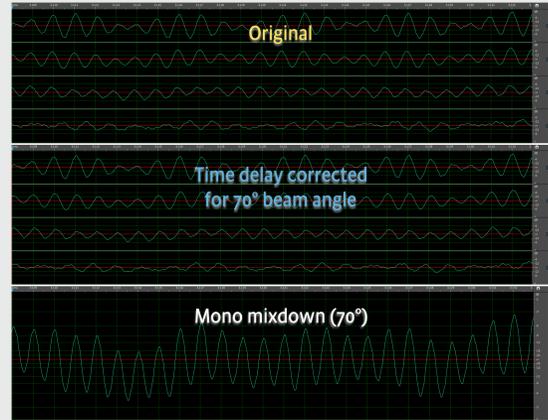
**Processing Test: Delay-Sum Beamforming**



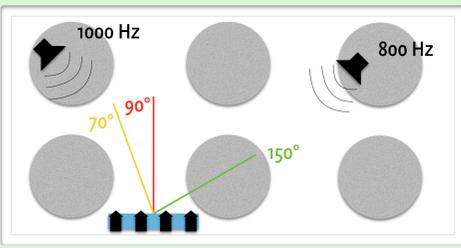
Audio recorded at 24-bit, 48 kHz into 4-channel WAV.



Individual tracks are phase-shifted to change direction of main array sensitivity by correcting time-delays.



**Test Results: Signal Separation**



Schematic illustration of test setup (see photo at left).



Mixdown of array output without beam-steering (default beam angle). 5 dB intensity difference means 1.5x loudness difference.

Beam steered toward 1000 Hz signal (70°). 15 dB intensity difference means on-axis signal appears three times louder.

Beam steered toward 800 Hz signal (150°). 23 dB intensity difference means on-axis signal appears almost five times louder.

**Tests show that a four-microphone array with 10 cm spacing allows for successful beam-steering to distinguish two spatially separated constant-frequency signals. Further tests are necessary to determine spatial resolution for variable-frequency signals like speech.**

**References**

Van Veen, B. D., & Buckley, K. M. (1988). Beamforming: A versatile approach to spatial filtering. IEEE ASSP Magazine, 5(2), 4-24.  
 McCowan, I. (2001). Microphone arrays: A tutorial. Queensland University, Australia.

Michel, U. (2006). History of acoustic beamforming. In Proceedings of the Berlin Beamforming Conference, Berlin, Germany.  
 Greensted, A. (n.d.). Delay Sum Beamforming – The Lab Book Pages. Retrieved January 03, 2018, from <http://www.labbookpages.co.uk/audio/beamforming/delaySum.html>