Flight Crew Audio Sound-Space

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Motivation/ Objective

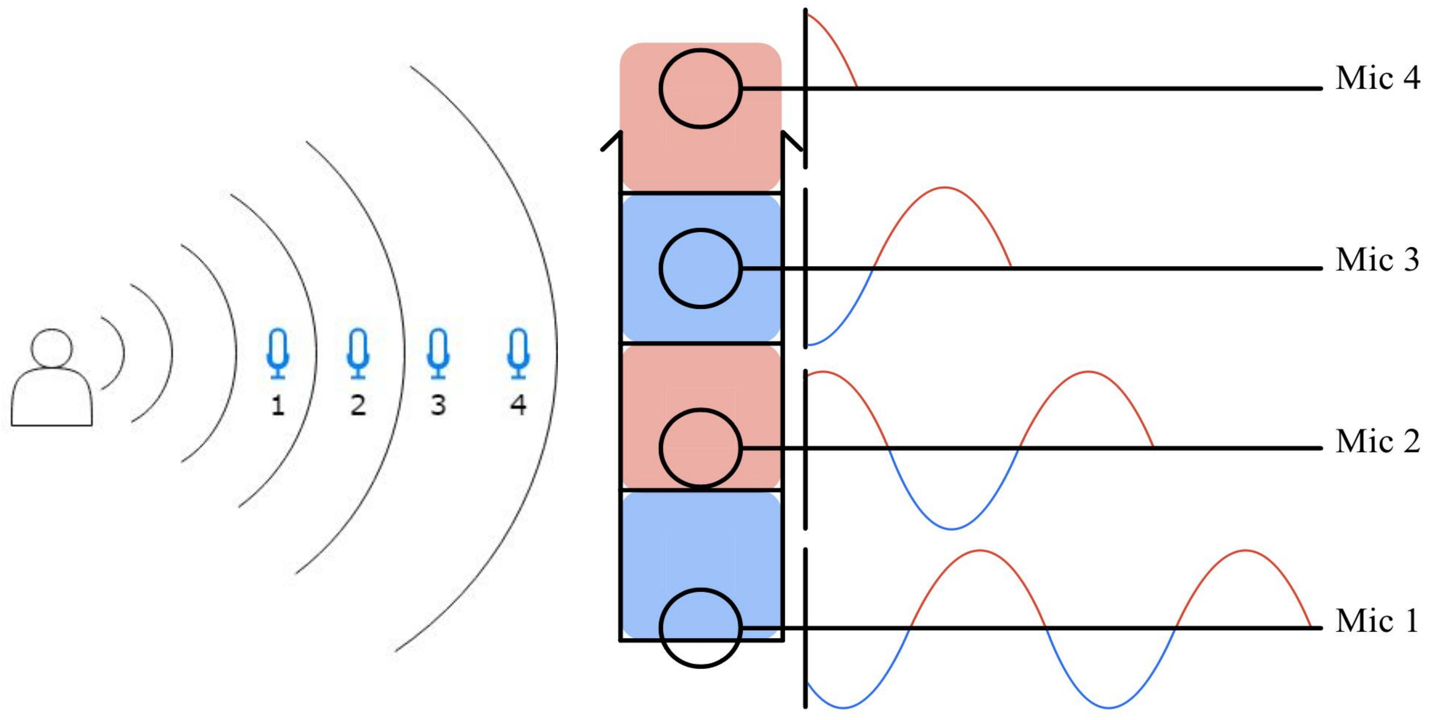
- Generating private sound fields at the Captain's and First Officer's nominal head position capable of replacing the headsets normally worn during normal flight operations (SoundSpace).
- Receiving clear flight crew spoken communications from the sound space equivalent to a near field Boom Microphone
- Generating 3D positional sounds (warning tones) within the SoundSpace

Requirements

- The sound space region shall be generated separately for the pilot and co-pilot
- Sound beams from different beamforming devices shall not interfere with each other's sound space regions
- The beamforming device shall be able to generate an area of audible sound encompassing the pilot's head and the co-pilot's head
- The audio processing unit should output a digital audio signal to the beamforming subsystem
- The beamforming device shall not generate more than 115 dB sound pressure levels in the ultrasonic range as per OSHA standards
- The beamforming device shall not generate more than 85 dB sound pressure levels in the acoustic range as per OSHA standards
- The sum of audio system delay shall not exceed 20ms as per RTCA-DO-214A
- The loudspeaker shall fit within a 20dB envelope over the frequency range of 350-6000Hz as per RTCA-DO-214A (2.3.2.1)
- The loudspeaker shall not exceed 10 percent at frequencies from 350-6000Hz as per RTCA-DO-214A (2.3.2.2)
- The beamforming device should pass necessary audibility tests in line with aviation standards
- The sampling frequency shall be 96kHz

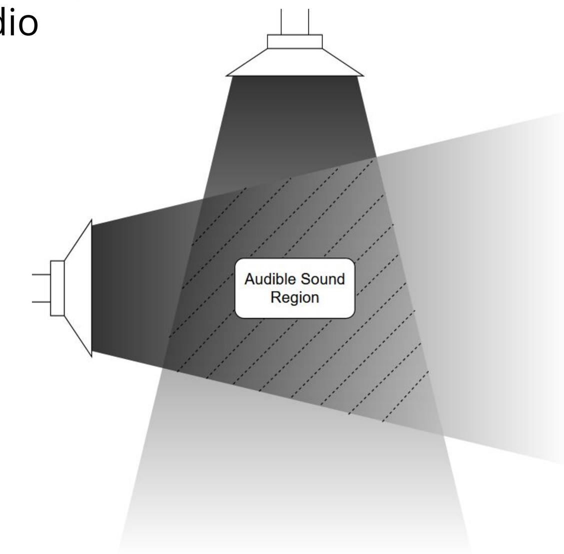
Far-Field Microphone Array

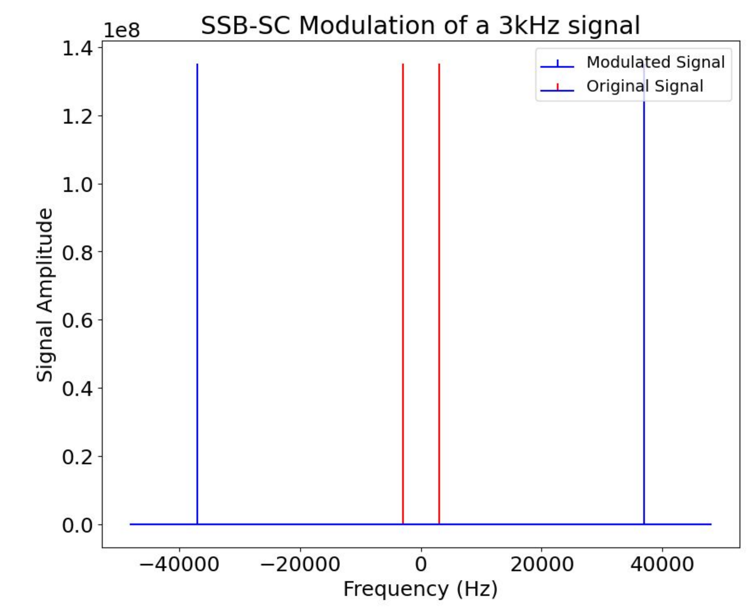
- The microphone array is in an end-fire configuration, where the microphones are placed in a line in the direction the sound is propagating and is most sensitive to sounds that are in the same direction as the array. In other words, the array enhances signals that are directly in line with the array.
- The array takes advantage of constructive interference, the signals detected by all the microphones are nearly identical, only delayed. By compensating for this delay and summing all the signals, the system can enhance the desired sound, the pilot or first captain's voice, and reduce background noise from other directions. Incoherent noise does not align when delayed the way a vocal signal does due to its randomness and lack of directionality.

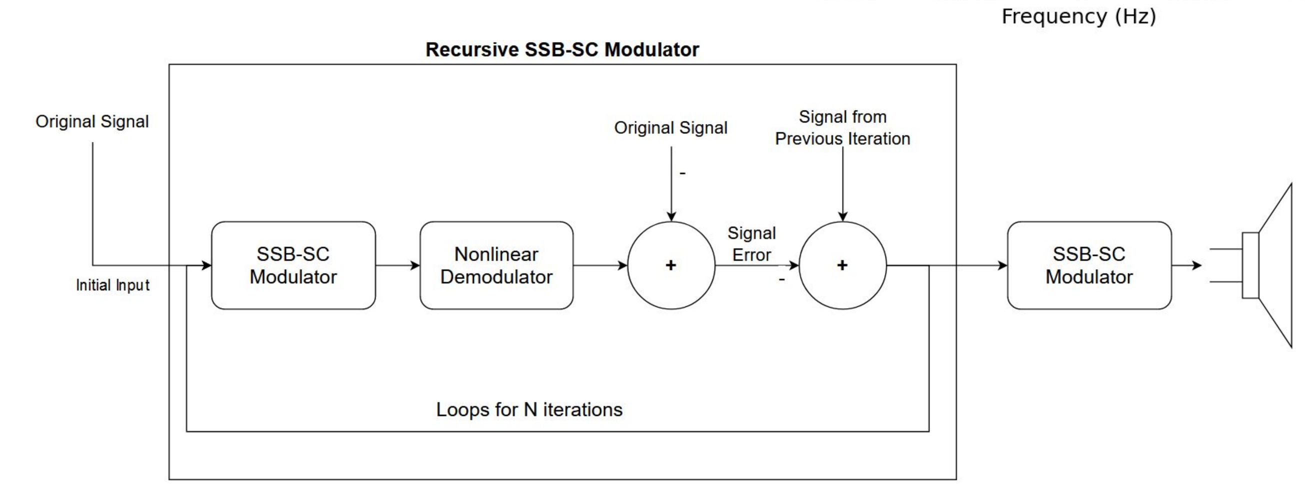


Ultrasonic Beamformer

- We use an octagonal array of ultrasonic transducers to produce audio "beams"
- The audio signal that will be sent to the transducer array is modulated using the recursive single-sideband suppressed carrier (SSB-SC) method
  - Introduces an error margin to help compensate for sound attenuating non-linearly in air
- An audible sound region is created through the intersection of two beamformers: one for emitting the carrier frequency at 40kHz and another for emitting the SSB-SC modulated audio

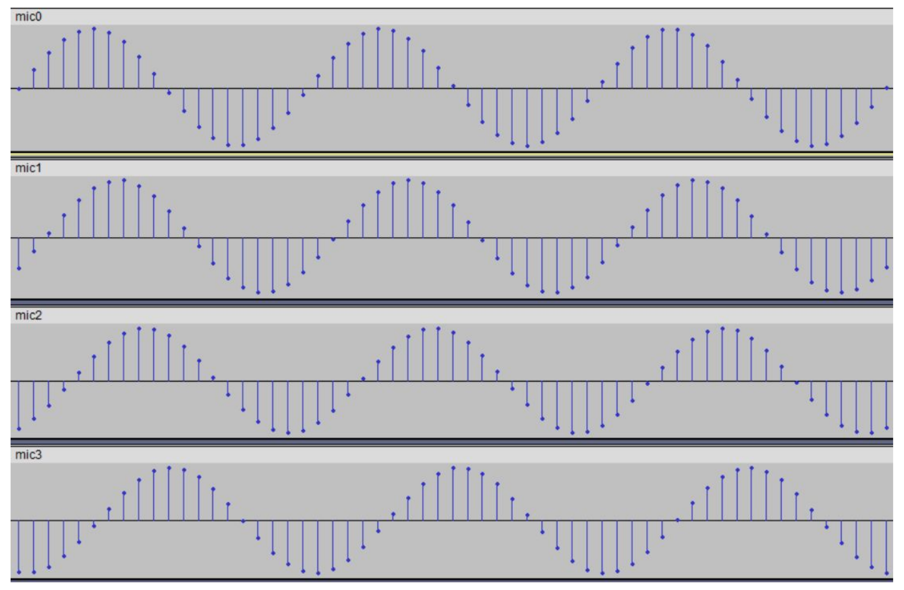


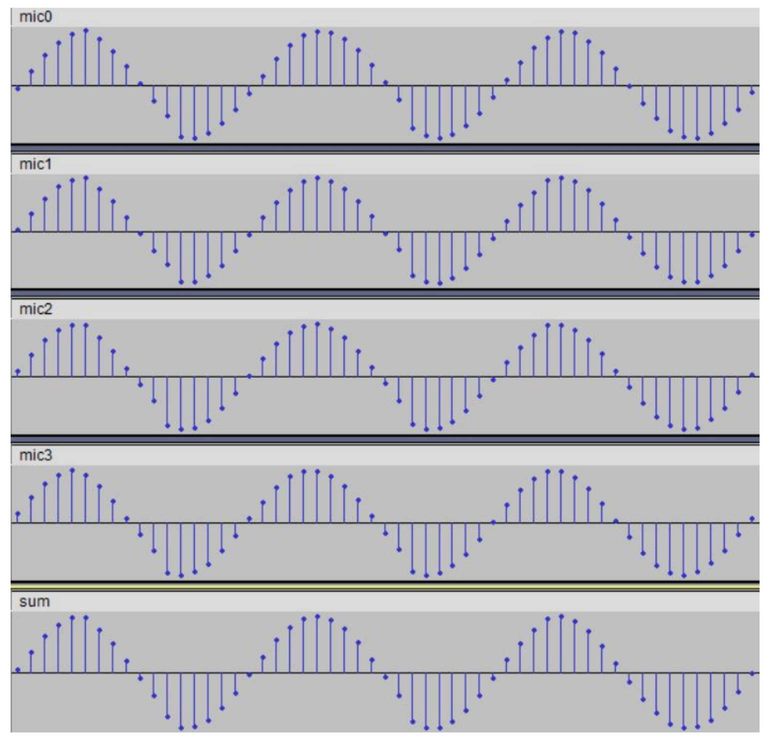




Far-Field Microphone Array Results

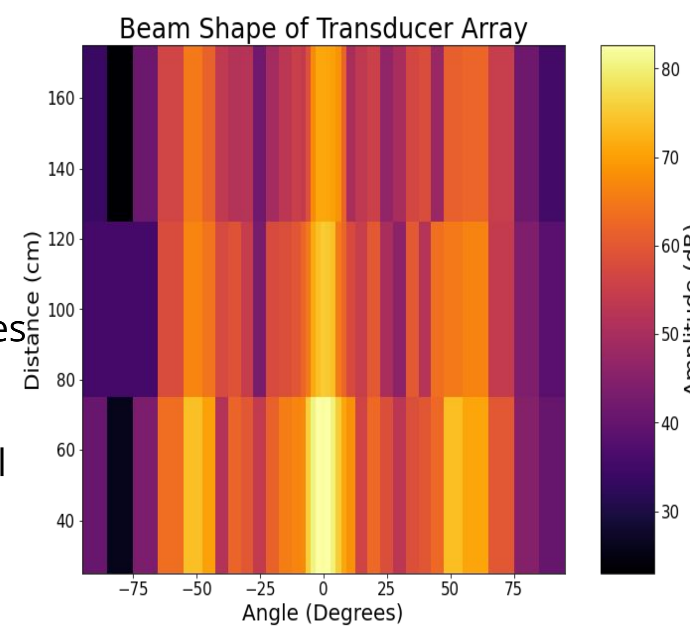
- The unprocessed 4-channel audio of a 1 kHz test tone shows the delay of each microphone, between each mic it is roughly 1 to 2 samples.
- After processing, the delay is compensated for. The 4 microphone channels almost perfectly align. The summed (5th) signal is an accurate representation of the test tone





Beamformer Results

- Conducted tests in an anechoic chamber to characterize
  - Intensity of ultrasound at locations
  - Effectiveness of demodulation techniques
  - Distortion and signal integrity
- Findings of anechoic chamber testing:
  - Suppressed Carrier Single Sideband modulation with two arrays at a 90° angle gives a more confined audible area
  - A 4 layered hexagonal aperture is the ideal configuration, and effects of adding additional layer is negligible
  - Sidelobes are present and audible at ±50°
  - Signal is slightly distorted by ultrasonic transducers



Future Work/Considerations

- More testing needs to be completed on:
  - Effects of different transducer spacing
  - Fine measurements to characterize the region of demodulation
  - Audibility of speech samples
  - Changing number of iterations of recursive modulation
- Parts of the design could be improved
  - Different Class D amplifiers to minimize distortion
  - More development of recursive modulation methods
- Improve real-time modulator:
  - Test other methods for implementing SSB-SC modulation
  - Verify the accuracy of the error margin calculated by the non-linear demodulator
- Performance of mic array could be improved
  - Increase sample rate of ADC to allow for more precision when delaying samples
  - Experiment with more complex microphone beamforming techniques
  - Increase number of microphones in array to improve voice isolation

Conclusion

- Through 2 quarters we were able to generate private sound fields of beamed audio in a localized soundspace using Suppressed Carrier Single Sideband and capture clear flight crew communications from the sound space using an array of far field mics.
- The beamforming array was theoretically difficult, and required many testing cycles. Through this, we targeted the ideal modulation method to create defined soundspaces.
- Implementing a far field microphone was also a challenge, but we were able to determine an ideal configuration and algorithm to reduce signal noise.

Acknowledgements and References

We would like to acknowledge John Reece, Todd Bolduc, Juan Pampin, James Ritcey, and Shanti Garman for their continued help and support throughout our project.

- Audio System Characteristics and Minimum Operational Performance Standards for Aircraft Audio Systems and Equipment." RTCA, 18-Dec-2013.

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