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## A Practical Comparison of Three Tetrahedral Ambisonic Microphones

Dan T. Hemingson<sup>1</sup> and Mark Sarisky<sup>2</sup>

Butler School of Music, The University of Texas at Austin, Austin, TX 78712, USA

<sup>1</sup>[dhemingson@mail.utexas.edu](mailto:dhemingson@mail.utexas.edu)

<sup>2</sup>[sarisky@mail.utexas.edu](mailto:sarisky@mail.utexas.edu)

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### ABSTRACT

This paper compares two low-cost tetrahedral ambisonic microphones, an experimental microphone and a Core Sound TetraMic with a Soundfield MKV or SPS422B serving as a standard for comparison. Recordings were made in natural environments, of live performances, and in a recording studio. The results of analytical and direct listening tests of these recordings are discussed in this paper. A description of the experimental microphone and the recording setup is included.

### 1. INTRODUCTION

The increased interest in surround-sound, be it for film, games, or recorded music, creates the need for an affordable and often portable means to experiment with and to capture surround soundscapes. The project described in this paper was inspired by a 5.1 surround-sound field-recording project using 3 Sennheiser MD421 and 2 MD431 microphones mapped to the dimensions of the speaker placement in the ACTLab Studio 4B in the School of Communications at the University of Texas at Austin. [1]

Ambisonic recording theory, techniques, and applications have been argued, written about, and practiced for a number of years since Michael Gerzon and Peter Craven designed the soundfield microphone. The further development of Ambisonics by Gerzon and Craven offers recording engineers the opportunity to make a single recording that can later be processed for monaural, stereo, and multi-speaker playback. A 3-dimensional playback can also be derived from the standard ambisonic B-format.

A tetrahedral ambisonic microphone consists of 4 cardioid microphone capsules mounted on the planes of

a tetrahedral array. The outputs of these capsules become the “A-format” signals that are matrixed to a “B-format” 4-channel signal. These channels are the equivalent of an omnidirectional output of the 4 capsules (“W”) plus 3 figure-of-8 microphones: one forward-facing (“X”), one left-facing (“Y”), and one upward-facing (“Z”). Recording engineers should feel comfortable with this resulting microphone arrangement, although usually minus the vertical microphone Z. The recorded B-format is then decoded for the number and placement of the playback speakers.

All of the listening samples used for this study were decoded with Visual Virtual Microphone (VVMic) by David McGriffy [2], a flexible program for use on a PC or Mac allowing from 1 to 32 outputs, and the version provided with the TetraMic allows for entering microphone calibration tables.

An advantage of the tetrahedral design is the single-point reference with all of the capsules being within the distance of a small tetrahedron. The time delay between the capsules in the tetrahedron is a minor factor. Another advantage is that, with all of the capsules contained within a single device, the microphone is more portable and the relative position of the capsules is fixed.

## 2. MICROPHONES

### 2.1. Experimental Microphone

The experimental microphone (Figure 1) is a “do-it-yourself” construction based on a design by Henry Walmsley. [3] Circuitry changes made initially to the microphone preamplifier design included changing the selection of the integrated circuits from TL074 quad-packages to the single op-amp TL071 devices, allowing each channel to be constructed in a modular style. This style would allow for a simpler conversion to a printed circuit board layout and a more compact preamplifier housing should the current device be rebuilt. The choice was also made to use a balanced output to allow for long cables between the microphone and the recording equipment. The line-level B-format outputs have successfully driven lines in excess of 80 meters in length. The microphone is powered by a pair of 9-Volt batteries, with several hours usage expected from a fresh set.

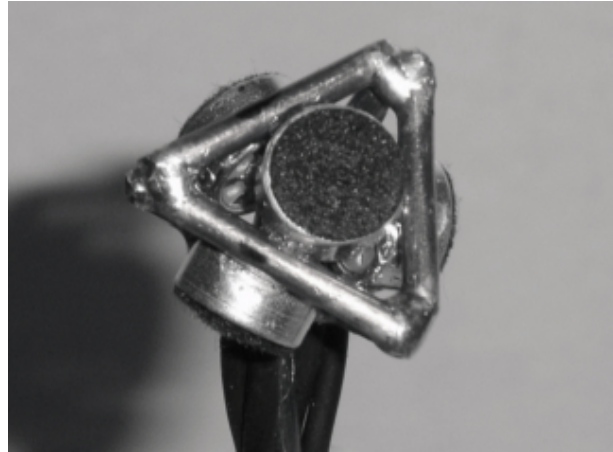


Figure 1 - Experimental Tetrahedral Microphone

There was no attempt to make frequency response corrections within the circuitry for the individual Panasonic WM-55A103 electret condenser capsules. The capsules were, however, selected for amplitude output matching at a single frequency out of a group of a dozen.

The first test was recording of an unwilling cat being carried in a circle around the device. Early tests with the microphone in outdoor locations proved that a windscreen was a requirement. Even a foam windscreen was often inadequate in moderate winds, but adding a furry windscreen made by WindCutter provided protection except on very blustery days.

A device for calibrating the amplitude of the four output channels was constructed from an earbud and two short pieces of plastic tubing (Figure 2), one end of the tubing stretched over the earbud and the other snugly slipped over one capsule at a time. Using a Swizz Army cable tester as a signal source for the earbud, the output of the corresponding preamplifier A-format output was set to an output of 1.5 Volts for the approximate 94dB tone.

Other amplitude values were used for calibration in early testing when very quiet or very loud environments are expected. Such was the case for a drum ensemble recording where the calibration was reset to a lower than normal gain. The equipment was setup and the author noticed static in the headphones. This was not

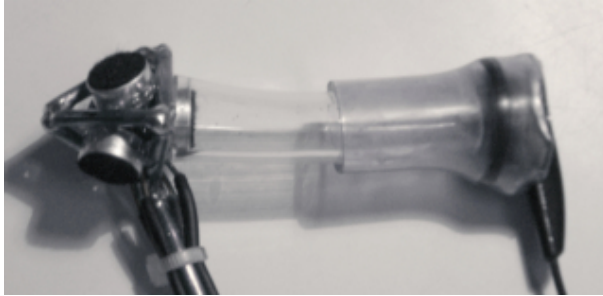


Figure 2 – Capsule calibration fixture

the first time this sound had been heard, but usually cleaning the connectors or a fresh set of batteries cured the problem, and indeed batteries seemed to fix the problem that time.

The recording was started and left unmonitored due to the location and unavailability of long lines to a remote location. The next day, the recoding was checked. The **W**-track (the sum of all of the capsule preamplifier outputs) was high quality, but the remaining 3 tracks contained an annoying hiss, all around different frequency groups. An oscilloscope connected to the capsule preamplifier outputs displayed oscillations, each channel in a different frequency band over the range of 700kHz-1.18MHz. The B-format matrix sum was adding and subtracting a mix of the oscillations in the **X**, **Y** and **Z** channels, resulting in the audible frequencies. A small capacitor was added to each channel preamplifier to roll off the radio-frequency range, and the noise floor for all of the microphone outputs decreased significantly.

An unresolved issue with the microphone is that of output muting in the presence of sounds louder than about 115dB, discovered prior to acquiring the furry windscreen and confirmed afterward when recording a fireworks display at close range.

The recording setup consists of a Focusrite Saffire Pro 26 i/o firewire interface connected to a MacBook Pro using Apple Logic. Mounted in an SKB StudioFlyer case, this system is truly portable with a recording time of more than 2 hours with a charged MacBook battery or it can be operated from AC mains with the computer power charger. A four-channel analog VU meter panel was added to facilitate gain-structure setting (Figure 3).

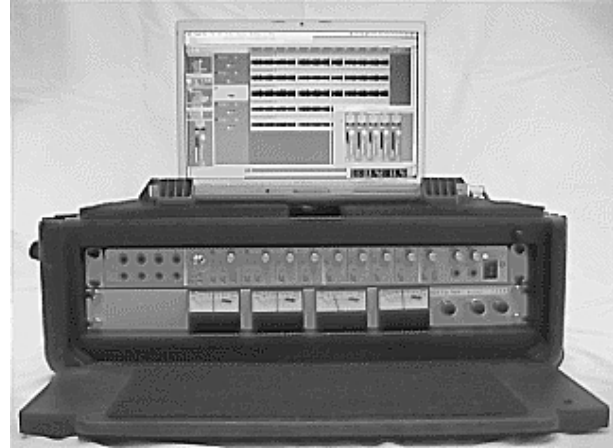


Figure 3 – Recording rig is self-contained for portable and battery-powered operation.

## 2.2. Core Sound TetraMic



Figure 4 – Core Sound TetraMic

About the time that the experimental microphone was constructed, Core Sound announced the TetraMic, the second relatively low-cost tetrahedral ambisonic microphone used in this comparison (Figure 4). The TetraMic outputs are A-format (four channels, one from each capsule, no matrix). The outputs from the microphone are unbalanced microphone-level, which experience has shown can pickup noise from a MacBook power adapter cable nearby. Furthermore, the output level of the capsules appears to be rather low based on the preamplifier gain settings, although testing such was not part of this comparison.

The optional TetraMic Phantom Power Adapters (four PPAs required) enable this microphone to be phantom powered from standard 48-Volt systems and convert the unbalanced PPA input to an XLR balanced output. Successful recording has been accomplished using more than 80 meters of cable from the PPA to the same Focusrite interface used for the experimental microphone, only using the mic inputs 5-8. Phantom power can be switched on from the Focusrite interface in banks of four. Core Sound offers both foam and furry windscreens, and both were used on the outdoor recordings. One complaint is that the output of the TetraMic is phase reversed; it remains uncorrected through the PPAs, so XLRF-XLRM cable adapters with pins 2 and 3 switched are required to record the A-format in-phase signals.

Core Sound states that the frequency response of the TetraMic when decoded with the included calibration tables is 30 Hz – 18.5 kHz +/- 2 dB. [4]

### 2.3 Soundfield MKV and SPS422B

The “standard” to which these low-cost microphones were compared comes from Soundfield. Actually, two models of the company’s products were used, the MKV, available only for recording in Bates Recital Hall at the Butler School of Music at The University of Texas at Austin, and the SPS422B, used for recording at the First Baptist Church of Austin. For use in this comparison, either is being considered as “the standard.” Both microphones have tetrahedral arrays, and both systems include a calibrated preamplifier/processor with, among others, line-level B-format outputs.

Recordings using the MKV were recorded via a Lynx sound card in a Mac computer using ProTools, while those using the SPS422B were recorded with an RME Fireface 800 interface connected to a Dell PC using Adobe Audition. The Soundfield microphone systems require AC power, limiting portability of the system to the limits of an extension power cord.

Soundfield states that both of these microphones have a “flat” response. [5]



Figure 5 – Soundfield SPS422B and experimental mic without its windscreen (left). Experimental mic with windscreen and Core Sound TetraMic on the dual mount (right).

## 3. RECORDING

Many of the early multiple microphone field recordings were made with the experimental microphone and the TetraMic on individual desk stands placed on the roof of a vehicle, and for indoor concert use with a Soundfield, each microphone was placed on a tall Shure microphone stand. A device was designed to mount the experimental and TetraMic on a single stand (Figure 5), and it was then possible to simultaneously record with all three microphones in a public performance space without causing consternation from the audience or performers.

Before each recording session adjusting the analog inputs on the Focusrite interface is a time-consuming task, not only for the 4-channels from each microphone, but the level differences between the microphones. The oscillator in the Swizz Army cable tester was used one channel at a time with a single analog VU meter to set

each of the 4 or 8 channels. Now to facilitate this adjustment, a cable has been built to insert the oscillator signal into either 4 line-level inputs or 4 mic-level inputs, while a switch on the VU meter panel selects output channels 1-4 or 5-8 of the firewire interface. The meters were selected from a group for accuracy at 0dB, and the interface output can easily be calibrated to within 0.1dB in the upper range of the meters.

Often in public performances, the recording levels were set based on experience and prayer or good luck. There was no opportunity for a sound check, and once the record button was clicked there was no chance to monitor progress since the Focusrite and computer were abandoned at the bottom of the microphone stand. Perhaps if there was an intermission, there might be time to confirm levels and to save files. There would never be time to save files and reset the input levels.

Perhaps one of the most important habits in recording is keeping good documentation for every recording and editing session. A standard file naming format and file locations, date and time, setup information, diagrams of microphone placement relative to the source, printed programs, and so on, have been critical to maintaining a smooth workflow. Detailed photographs have also proven to be valuable, especially when recording with all 3 tetrahedral microphones.

#### 4. LISTENER SURVEY

The author has demonstrated ambisonic playback in the ACTLab studio both in horizontal 2-dimensional and 3-dimensional playback arrangements. The in-house system is arranged in the traditional rectangular 5.1 arrangement with the HF boxes flown from a lighting grid about 4 meters above the floor. The center front HF speaker was not used for the ambisonic demonstrations. For 3D playback, the author installed additional speakers at 1 meter above the floor, one below each of the 4 flown corner speakers. The audience came from a wide mix of academic disciplines, and comments were more toward the overall project rather than toward the microphones used for the recording.

To specifically avoid the group survey method, the author chose a select group of recording engineers, faculty members, and musicians to listen to samples of recordings in an environment and on equipment with which they were familiar; that generally implies their home system. Very purposefully, this returned

subjective responses based on consumer listening situations, not just evaluations from well treated and tuned studio control rooms. Most home systems are not surround equipped, but “home system” does not imply low quality.

All listeners were sent a cover sheet with a short description of the recorded material. The letter stated, “There are no specific questions. I am interested in what you hear (or don’t).” This was not to be a survey of the selections, genre, or performance. They were asked not to let prior knowledge of any microphone sway their response and that the length and detail of their response was up to each listener. Yes, it would be subjective. Two final sample discs were distributed, each to different survey groups.

The first group was given a CD with a recording from a public performance by the University of Texas Chamber Winds ensemble. No equalization was applied to this recording. The only processing was to normalize the tracks. The hour-long sample contained only the omnidirectional **W**-channel from the B-format files, and, in a quasi-random fashion, cut into 38 tracks, switching from one microphone to another in no predictable order or time frame. The cuts occurred during room ambience, applause, and at musically appropriate times while the chamber group was playing. The purpose of a monaural playback was to eliminate distraction from any surround localization anomalies in the surround recordings.

The second disc contained a recording of the 56-rank Casavant Organ in the Sanctuary of the First Baptist Church of Austin. The material was one selection just over 2 minutes in length heard from each of the 3 microphones from a recording session played by the church organist. The disc was available to the listener as either a stereo CD or a DVD in surround 5.1 with no audio on the front center channel.

The tracks from the Core Sound and Soundfield microphones were not equalized after decoding for the second disc. However, Core Sound provides calibrated equalization curves that are plugged into the VVMic decoder software and the Soundfield processor contains equalization curves. To level the playing field, EQ was applied to the experimental tracks on disc two: a slope from 0dB at 400Hz to +10db at 50Hz, then shelved at +10db to 16Hz.



## 5. SURVEY RESULTS

Listener comments were, of course, all formatted in their own way. The author has attempted to consolidate and paraphrase them fairly. Similar comments are listed only once.

### 5.1. Soundfield MKV and 422B:

- low frequency response as good as the TetraMic, but the “attacks” were not as prominent
- lows were “gruff”
- like the deep, darker sounds
- lows are “fuller”
- extra “boomy” bass
- could “feel” low notes as well as hear them
- better mid-range response than TetraMic; melody was better balanced with harmony and bass lines
- too bright, too open
- highs clearer
- overall brighter and crisper
- clarinets and flutes “a little edgy”
- instruments easily distinguished
- picks up “small sounds” better
- front imaging very good, rear imaging even better
- reverb at end of organ piece had an audible image
- “sounded like I was in a large auditorium”
- sense of “space”
- as ambient as the TetraMic, but has more emphasis on low-mid frequencies
- would like bit more “attack” in mid-range
- first choice

### 5.2. Core Sound TetraMic

- better frequency response than experimental
- warm sound, but muffled
- not as open as others
- sharper sound, but less depth
- mid-range didn’t have same prominence as the experimental
- instruments seem to run together, not as distinguishable
- not very crisp, undistinguished melody
- detected “hiss”
- no obvious noise
- closer in performance to the experimental than to the MKV
- front imaging very good, rear much better

- more ambient than experimental, especially in lowest frequencies
- not deficient in low frequencies
- suspicious of mid-range sound

### 5.3. Experimental Microphone

- deficient in low-frequency band
- sound is “flat”, both in frequency response and tonal quality
- good mid-range presence
- individual instruments and sound filled the room
- low and high instruments better than other mics
- no obvious noise
- front imaging very good, rear imaging not as good
- sounds “natural”
- hear total range of pitches
- sounds like a lot of organs in churches which don’t have much ambience
- realistic representation of the original performance
- first choice

### 5.4 Frequency analysis graphs

One listener included frequency analysis curves of all 3 microphones (Figures 6-8). He had conducted a self-imposed blind test and did not know the order of the microphones on the organ survey CD. He reported, “[I] expected to see considerable differences [on all 3 microphones] at extreme lows and highs... and found none to speak of.”

### 5.5 Conclusion

Overall favorite ratings from the listeners were split evenly between the Soundfield and the experimental microphones. One sent in an undetermined vote, saying he could never decide between the Soundfield and the experimental devices. The author is simply reporting the choices and did not take sides.

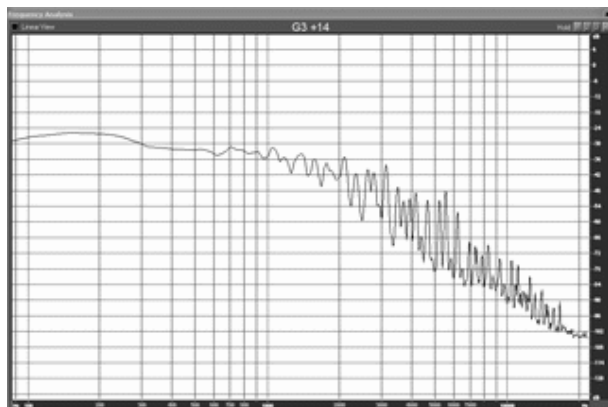


Figure 6 – Log curve of the experimental microphone \*

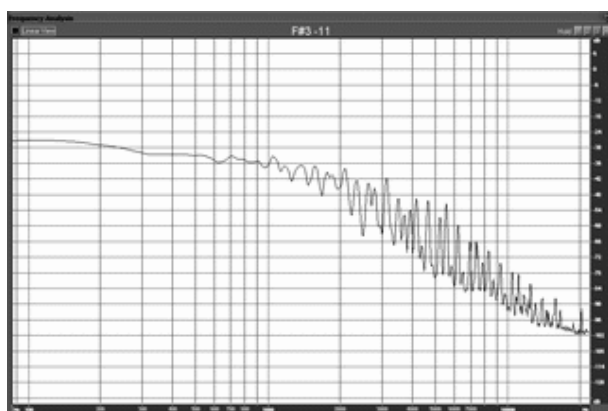


Figure 7 – Log curve of the Core Sound TetraMic \*

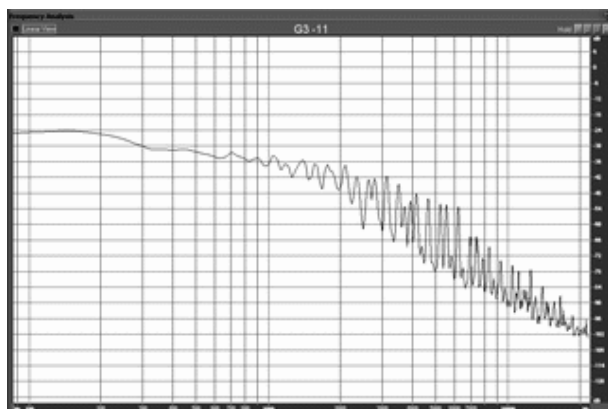


Figure 8 – Log curve of the Soundfield SPS422B \*

\*Frequency analysis plots courtesy of John N. Eddins, Jr., Dynastat, Inc., Austin, TX, USA.

## 6. ACKNOWLEDGEMENTS

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Thanks to David Weigle, organist at the First Baptist Church of Austin, for the use of a selection from his recording session and to the First Baptist of Austin, Austin, TX, for the use of their sanctuary.

Thanks to the listeners who accumulated many hours evaluating the recordings.

## 7. FUTURE WORK

A wish list has been established for the experimental microphone:

- Complete the work planned for the anechoic chamber, including frequency response, impulse, and localization testing of all 3 microphones
- Easier way to setup/confirm capsule gain
- A-format output connectors for setup purposes
- Layout capsule preamplifier p.c. boards
- Phantom power capabilities

## 8. REFERENCES

- [1] <http://actlab.us/actlab/danh/Dream/Page31.html>
- [2] [www.vvaudio.com/products/VVMic/](http://www.vvaudio.com/products/VVMic/)
- [3] [www.interestingelectronics.com/old/henrys\\_interesting\\_electronics/cheap\\_soundfield/cheap\\_soundfield.htm](http://www.interestingelectronics.com/old/henrys_interesting_electronics/cheap_soundfield/cheap_soundfield.htm)
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NOTE: Typographical corrections were made to the original paper and are included on this version of the paper on May 28, 2009. Reference section URLs updated November 24, 2015.