

## ► Design and Measurement Of a Dipole Microphone

By Mark Williamsen

Tone burst techniques, gated measurement, analog computing, and imaginary numbers.

I've been interested in microphones for some time, especially those used to record music in concert and studio settings. More recently, however, I've become especially interested in how various microphones differ, how to measure and compare those differences, and what design trade-offs go into the development of a new microphone. This article examines a particular kind of microphone, the acoustic dipole or "gradient" microphone and its design parameters<sup>1</sup>.

First I'll describe its operation mathematically in enough detail to predict frequency and polar response characteristics in a spreadsheet. Next, I'll discuss the techniques needed to obtain accurate measurements of microphone frequency and polar response, using a computer sound card and some simple software programs. The spreadsheet and computer source code are posted online ([www.audioXpress.com](http://www.audioXpress.com)) for readers to download. Finally, I'll show the construction of a prototype dipole microphone using two omnidirectional electret condenser microphone cartridges.

My goal here is not to produce a design that's better or different from existing designs, but rather to gain a more complete understanding of what goes into microphone design and measurement. This way, we can all be a little more confident about making the best use of the microphones that are available to us.

### MICROPHONE TYPES

The simplest microphone is the omnidirectional pressure microphone, which senses instantaneous pressure at a point in space. Pressure doesn't have a direc-

tion, so an ideal pressure microphone responds equally to sounds from all directions. Sounds arriving simultaneously from various directions simply add up to a single pressure value at the location of the microphone. Note, however, that the addition is instantaneous, so phase matters when doing the summation. Two sounds at the same frequency could add up constructively or destructively, depending on their phase relationship.

Now suppose you want to design a microphone that picks up sounds along a particular axis, while ignoring sounds along some other axis. A simple technique is to place two pressure microphones of equal sensitivity near each other, and then obtain the difference between the two signals. Sound waves traveling along the axis of the dipole (that is, along a line through the acoustic centers of both microphones) will arrive at slightly different times. While the amplitudes will be roughly the same, there will be a phase shift between the two signals so they don't completely cancel. This, then, is the "on-axis" signal of the directional microphone.

Sound waves arriving from any direction in a plane perpendicular to the dipole axis will arrive at both capsules at the same time, so there will be no phase shift between them, allowing for complete cancellation of the signal. This is the "off-axis" response, for what you now see is a bidirectional microphone, having a figure-eight polar response pattern. The combined dipole microphone now responds to the gradient of pressure instead of pressure itself, and has a preferred axis

for pickup.

But what if you want to pick up sounds arriving from one direction along the dipole axis while rejecting sounds arriving from the opposite direction? It turns out that the dipole gradient microphone can distinguish front from back, in the sense that sound from the front gives a gradient signal which is in phase with the omni response, while sound from the back gives a gradient signal which is opposite in phase to the omni response. Assuming the omni and on-axis gradient responses have the same sensitivity, they may be added together to obtain a response that is twice the omni response on-axis from the front, equal to the omni response from the side, and null (complete cancellation) along the dipole axis from the rear.

The dipole microphone is now giving what is commonly called a "first-order cardioid" polar response pattern, preferring sounds from one direction while rejecting sounds from the opposite direction. This microphone type is widely used—both in studios and in live recording—because it allows direct sound from a performer or instrument to be emphasized, while reducing ambient and reverberant sounds<sup>2</sup>.

Most commercial microphones having a first-order cardioid response actually have just one diaphragm and one electrical output. The gradient is obtained by having two ports to accept sound, which is then routed to the front and back of the diaphragm with appropriate time delays and other acoustic elements. This is a form of analog computer, implemented with air passages. The best way to see

how this works is to take one apart, although you may ruin it in the process.

For my work here, I'll use two separate omnidirectional capsules and then develop the various polar patterns using an analog computer implemented with gain stages, adders, and an integrator. Clearly, the computations could also be implemented in a digital computer, but real-time digital signal processing is beyond the scope of this article<sup>3</sup>.

## MATHEMATICAL DESCRIPTION

Now let's work out the math needed to describe the polar and frequency response for the dipole microphone. First you need some assumptions, so things don't get completely out of hand. I'll assume the dipole is made of two ideal pressure microphones having identical sensitivity, and flat frequency and phase response over the range of interest. I'll limit range of interest to 100Hz to 10kHz, for reasons that will become clear as we go along. I'll assume free-field conditions, considering only the direct sound from an ideal point source, with no reflections or reverberations, and, of course, no ambient noise.

Sounds from the ideal point source will propagate as spherical waves, so assuming no transmission loss in the medium, acoustic pressure will fall as the inverse of distance from the source. Acoustic power, of course, falls as the inverse of distance squared, but the microphone responds only to pressure, not power, thus the 1/r dependence. The central axis of the dipole will, for my purposes, always remain in a plane that includes the point source.

Now I write the wave equation giving instantaneous pressure  $p$  at distance  $x$  and time  $t$ , for a steady-state sound of frequency  $\omega$  in units of radians per second<sup>4</sup>:

$$p(x, t) = \frac{X_0}{x} e^{i(kx - \omega t)} \quad (1)$$

Here  $e$  is the base of natural logarithms (2.71828...), and  $k$  is the wave number  $2\pi/\lambda$ , where  $\lambda$  is the wavelength of sound at the given frequency. I define a reference distance  $x_0$  at which pressure is arbitrarily set to unity. This will be the distance from the point source to the acoustic center of the dipole. Because wavelength is just  $c/f$ , where  $c$  is the speed of sound and  $f$  is the frequency in units of Hz, you can now write the wave equation as<sup>5</sup>:

$$p(x, t) = \frac{X_0}{x} e^{i2\pi f(x/c - t)} \quad (2)$$

You've probably noticed that I'm using complex notation here, where  $i$  is the square root of  $-1$ . It may seem like an unnecessary burden, but it actually helps to keep track of phase when you do the calculations in Excel or other math programs. You just need to be sure to let Excel know that these are complex numbers. Note that some math programs expect "j" instead of "i" when entering imaginary values.

Now consider an angle  $\theta$  between the axis of the dipole and a line from the point source to the center of the dipole, where  $d$  is the acoustic distance between the two microphone capsules in the dipole (Fig. 1). By the law of cosines, the distance from the source to each microphone capsule is given as<sup>6</sup>:

$$x_A = \sqrt{x_0^2 + \left(\frac{d}{2}\right)^2 - 2x_0\left(\frac{d}{2}\right)\cos\theta} \quad (3a)$$

$$x_B = \sqrt{x_0^2 + \left(\frac{d}{2}\right)^2 + 2x_0\left(\frac{d}{2}\right)\cos\theta} \quad (3b)$$

Here I've labeled the distance from source to microphone as  $x_A$  for the front capsule and  $x_B$  for the rear capsule. For large  $x_0$  (that is,  $x_0 \gg d/2$ ), this can be simplified to  $x_0 \pm (d/2)\cos\theta$ , but let's keep the full form for now because I will be considering small  $x_0$  (that is,  $x_0$  comparable to  $d/2$ ) when I look into the proximity effect.

Now you can construct a spreadsheet to calculate the omni, gradient, and

cardioid polar response patterns for an acoustic sine wave. I'll begin with some columns for given values, including the speed of sound (345m/sec assuming 22°C ambient temperature), distance from source to dipole (1m), length of the dipole (0.01m), frequency of the incident sound wave (1000Hz), and angle of dipole axis with respect to the source (0°). I've used the "Fill Down" command to copy the given values into 72 consecutive rows, except for angle, which I set to run from 0° to 355° in 5° steps. I placed a column alongside the angle which converts from degrees to radians using the Excel built-in function RADIANS(), because many Excel functions assume radians as input. Now you calculate the distance from source to microphone A using equation (3a), and from source to microphone B using equation (3b).

All calculations up to this point have been real numbers, but now I will switch over to complex math to obtain the response to the source signal using equation (2) for each capsule, replacing  $x$  with the value of  $x_A$  or  $x_B$  as needed for each capsule. Because I've assumed steady-state conditions, the phase relationship between the two capsules will be the same at any time  $t$ , so you can set  $t = 0$  to simplify the calculations. You then take the complex sum and complex difference of the signals from capsules A and B. The absolute value (radius in the complex plane) is the output level in units of pressure where the reference level is unity, while the argument (phase angle in the complex plane) is the output phase in radians. A radar plot of the difference output level should show a figure-eight pattern, while the sum output level should be a circle.

Use the Excel functions IMABS() and

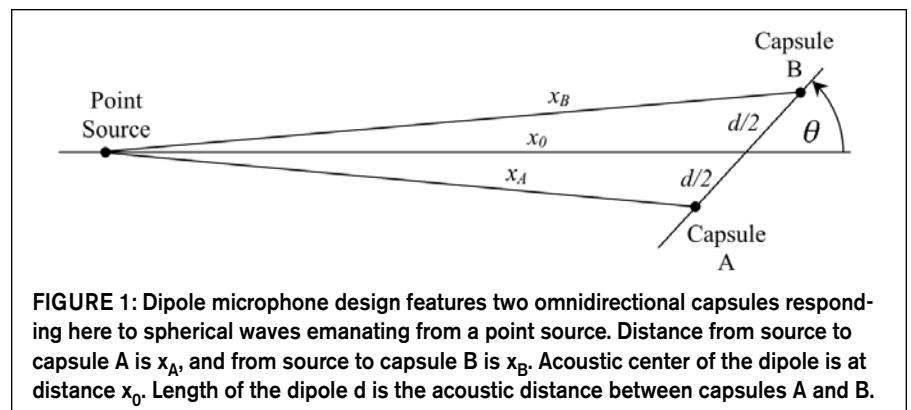


FIGURE 1: Dipole microphone design features two omnidirectional capsules responding here to spherical waves emanating from a point source. Distance from source to capsule A is  $x_A$ , and from source to capsule B is  $x_B$ . Acoustic center of the dipole is at distance  $x_0$ . Length of the dipole  $d$  is the acoustic distance between capsules A and B.

IMARGUMENT() to obtain real-valued numbers for plotting. If Excel gives a “#NAME?” error when you try to use complex math functions, check to be sure that you have the Analysis ToolPak enabled, under “Tools Add-Ins. . .”

But now you notice that at 1kHz, the difference (gradient) output level is much lower than the sum (omni) output level, and as you change frequency you find that the difference output level has a large frequency dependence, increasing along with frequency. What happened was that when I calculated the difference between the microphone A and microphone B signals, I effectively took the differential  $\Delta p$ , which caused a factor of  $i(2\pi f/c)\Delta x$  to pop out, where  $\Delta x$  is just the dipole length  $d$ . This factor causes the frequency dependence along with a  $90^\circ$  phase shift.

To make the on-axis difference signal look like the omni signal, you must divide out a factor of  $i(2\pi f/c)\Delta x$ . Luckily, you can integrate the gradient signal with respect to time to remove  $i2\pi f$ , and  $\Delta x/c$  is just a constant gain factor equal to the dipole length divided by the speed of sound. So add a new column to the spreadsheet where these factors have been divided out of the complex difference response.

With this change the gradient response is now uniform over frequency, but still off by a factor of 2, due to the sum signal including the signal from two capsules. So you divide the complex sum response by 2 (think of it as an average), and can now compare the amplitudes

of the omni and gradient responses on an equal basis. Alert readers will have noticed that  $x$  appears twice in equation (2), but when I integrated the difference signal with respect to time, I accounted for only one of the  $x$ 's. The presence of this second  $x$  is actually the origin of the proximity effect in gradient microphones.

The final step is to show the cardioid response in the polar plot. This is done by taking the complex average of the omni and gradient responses and plotting the absolute value. The result should be a familiar heart-shaped pattern, but it's pointed in the wrong direction!

It turns out that I overlooked a minus sign when obtaining the time integral of the gradient response. Reinstating the minus sign will point the cardioid microphone in the right direction. So you can obtain both a front-facing and rear-facing cardioid response from the same omni and gradient signals (Fig. 2).

By the way, an important characteristic of first-order cardioid microphones is that the response at  $90^\circ$  and  $270^\circ$  should be exactly 50%, or  $-6\text{dB}$  when plotted logarithmically. You now have a toy model that reveals many of the characteristics of real microphones, as frequency and distance from the source vary. If you are using Microsoft Excel, you can download a fully populated spreadsheet of this model ([www.audioXpress.com](http://www.audioXpress.com)).

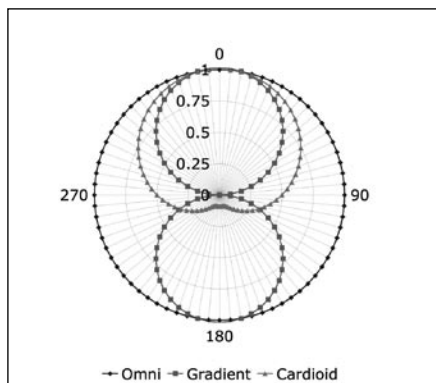
## MEASUREMENT TECHNIQUES

To test the model spreadsheet, I'll describe the construction of a prototype dipole microphone. But first, you need a set

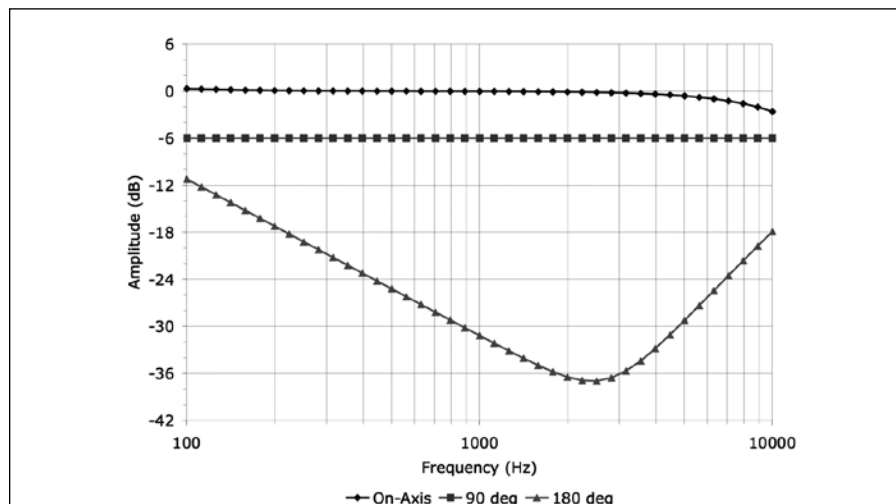
of measurement techniques in order to evaluate the prototype. According to the assumptions that went into the spreadsheet, you want to measure the free-field steady-state response to spherical sound waves arriving from a point source, in the frequency range from 100Hz to 10kHz. If you are already set up to measure microphone polar and frequency response, you can skim over this part of the article. For readers who are interested in developing their own setup for microphone measurements, however, I'll give the complete details.

To achieve free-field conditions is a bit of work, but well worth the effort. Building an anechoic room is one option, but way beyond reach for most of us. If you want to be able to stand up inside your anechoic chamber, you'll need 2m clearance vertically, plus 2m above and 2m below for the sound-absorbing wedges, requiring a space of at least 6m in height for construction. This would be for a room with a cutoff frequency around 86Hz, where 2m is a half-wavelength. Below cutoff, the chamber reverts to pressure mode and is no longer anechoic. Readers living in a mild climate might consider using an open field or empty parking lot, especially if they can dig a small pit for a loudspeaker to be mounted so the baffle is even with the ground, pointing straight up in a  $2\pi$  (hemispherical) configuration.

The microphone under test can then be suspended directly above the loudspeaker, pointing straight down. Note, however, that you'll need to accept what-



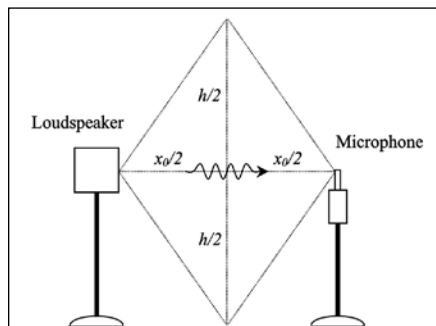
**FIGURE 2A:** Calculated polar plot of 300Hz response with dipole length of 10mm. Signals shown include omni, gradient, and cardioid. Distance from source is set to 1m, so proximity effect is minimal. Radial axis is linear.



**FIGURE 2B:** Calculated frequency response of microphone with dipole length of 10mm. On-axis,  $90^\circ$ , and  $180^\circ$  plots are shown. Vertical axis is in dB.

ever ambient noise is present in the environment. Also, the speed of sound has a rather strong temperature dependence, varying almost 4% over the temperature range from 0 to 20° C.

I actually chose to obtain free-field conditions in my living room with a tone-burst source and gated measurement system<sup>7</sup> using wave files and the sound hardware of a laptop computer. My living room ceiling has a height of 2.43m. With loudspeaker and microphone suspended 1m apart midway between floor and ceiling, the direct sound arrives at 2.9msec after launch, while the first echoes arrive at 7.6msec (Fig. 3). That gives me 4.7msec of direct sound with no echoes, satisfying the requirements for a free-field measurement. I can fit at least one complete cycle of a free-field measurement at frequencies of



**FIGURE 3:** Direct sound arrives at the microphone at time  $t = x_0/c$ , where  $x_0$  is the distance from source to microphone, and  $c$  is the speed of sound in air. First echo arrives a bit later, after being reflected off floor and ceiling at height  $h$ . Free-field conditions are obtained during the interval between direct sound and first echo<sup>7</sup>.

212Hz and above.

When I measure frequencies below 212Hz, at least some of the first echo from floor and ceiling is included in the measurement. If loudspeaker and microphone are moved closer, to 0.3m separation, then I have 6.2msec in the free-field regime, allowing good measurements down to 161Hz. Measurements below cutoff don't blow up, but rather begin to include more and more sound from directions other than the source with falling frequency. While it's definitely not steady-state, you will see that this technique gives results that are easily good enough to justify the effort in obtaining them.

All test signals were calculated and

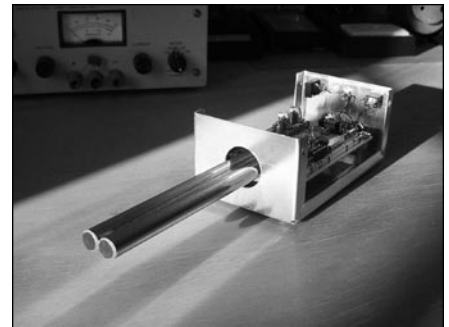
written directly to wave files using a simple command-line utility program called "tbg.exe," written for the purpose in C++. This technique of using offline signal generation and analysis has worked out very well for me, and seems to have no real disadvantages when compared to more sophisticated graphical user interface (GUI) programs that make their measurements in real time. Complete source code is available for readers to download ([www.audioXpress.com](http://www.audioXpress.com)).

## PROTOTYPE CONSTRUCTION

Now you can build a prototype dipole microphone, to see whether the characteristics predicted in the model spreadsheet are meaningful. The first thing you must decide on is the length of the dipole. As you can see from the spreadsheet model, there are nulls (complete cancellation) in the on-axis omni response whenever the dipole length is equal to odd multiples of a half wavelength. For instance, a dipole length of 10cm gives nulls at 1725Hz, 5175Hz, 8625Hz, and so on. I've used 10mm diameter capsules for my prototype, so the closest I can easily place the acoustic centers of the capsules is 10mm, giving the first null at 17.25kHz. This helps to explain why, especially for large diameter microphones, it may be better to avoid gradient behavior at high frequencies.

The basic requirement is to sample pressure at two nearby points in space. To do this I used a pair of Panasonic omnidirectional electret condenser microphone cartridges, part #WM-034BY, which I had on hand. It seems that this part number has become obsolete, but the cartridges themselves are completely generic, and can be easily substituted without affecting performance. I chose to mount the microphone capsules facing upwards in a probe configuration (Photo 2A). This allowed me to put the electronics beneath the microphone, away from the line-of-sight for incident sounds, and made the microphone nearly symmetric about the vertical axis for polar plots.

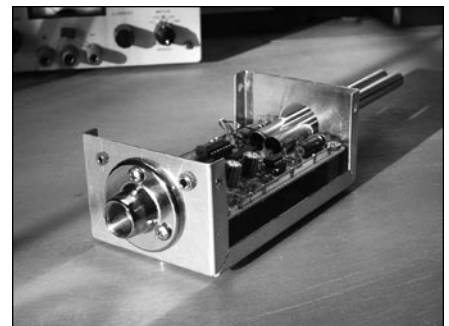
You may ask why I didn't face the capsules forwards, or perhaps one forward and the other one backwards. I actually considered all of these configurations, but didn't try them because I had in mind to make a bidirectional microphone that would allow me to develop both front-



**PHOTO 2A:** Two electret microphone cartridges are mounted side-by-side in a probe configuration above an electronics package. Thin wall brass tubes are tack-soldered to each other, and rigidly mounted to the perfboard circuit assembly.

facing and rear-facing cardioid patterns from the same signals. I also considered the fact that gradient microphones are notoriously sensitive to handling noise and mechanical vibration. Facing both capsules in the same direction should minimize the difference signal between them when both are subjected to the same vibrations<sup>8</sup>.

As shown in Photo 2B, two brass tubes hold the capsules up above an electronics package contained in an aluminum project box. The tubes and project box are all tied to ground for proper shielding. I obtained a threaded mike stand adapter from Atlas Sound which I mounted on the bottom of the project box, centered along the vertical axis of the microphone. Although not shown, you can easily fit an inexpensive foam windscreen around the



**PHOTO 2B:** Threaded mike stand adapter is aligned with the acoustic center of the dipole. Power supply input and line level output jacks are also mounted on the bottom end of the project box.

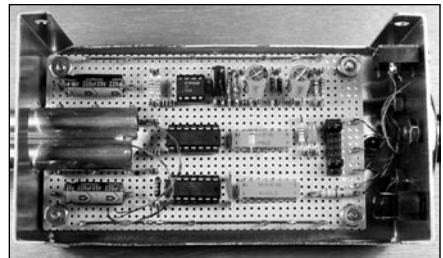
capsules if needed.

I provided two jacks at the bottom of the project box, one for a 9 to 12V DC power input, and one for a stereo

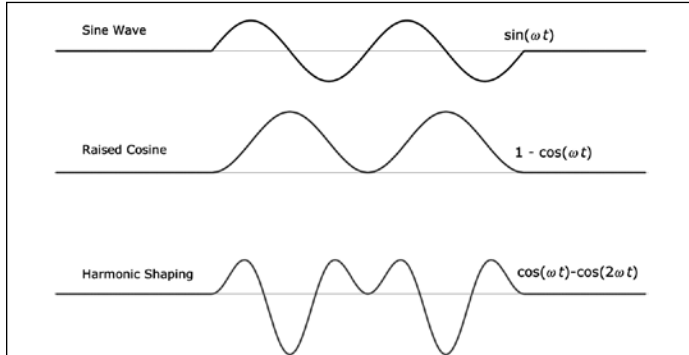
line level output to match my computer's sound card input. The two capsules should be rigidly mounted with respect to each other, and should have mechanical resonances well damped. In my prototype, I found that the wires connecting to the capsules had a strong resonance in the low audio range, so I stuffed the brass tubes with tissue paper to keep the wires quiet. I also experimented with using a pyramid, glued from foam-core board, to blend the surfaces of the project box to the probe tubes, but wound up not using it for the measurements in this article.

The electronics package implements the analog computer I designed while coding the spreadsheet earlier in this article. It consists of gain stages, adders, and an integrator, as indicated in the block diagram of Fig. 5. A detailed schematic is given in Fig. 6. The completed assembly is in Photo 3.

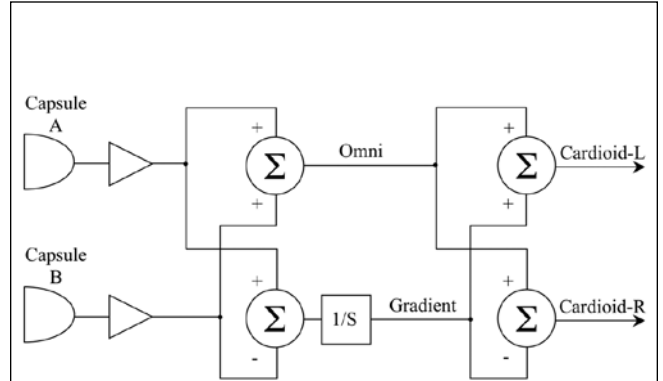
I mounted all parts on a piece of 0.1" grid perfboard using DIP sockets and 0.042" perf-clips. Wiring was done point-to-point using a "wrap and burn" technique with enameled magnet wire. A set of jumpers JMP1 allows me to re-



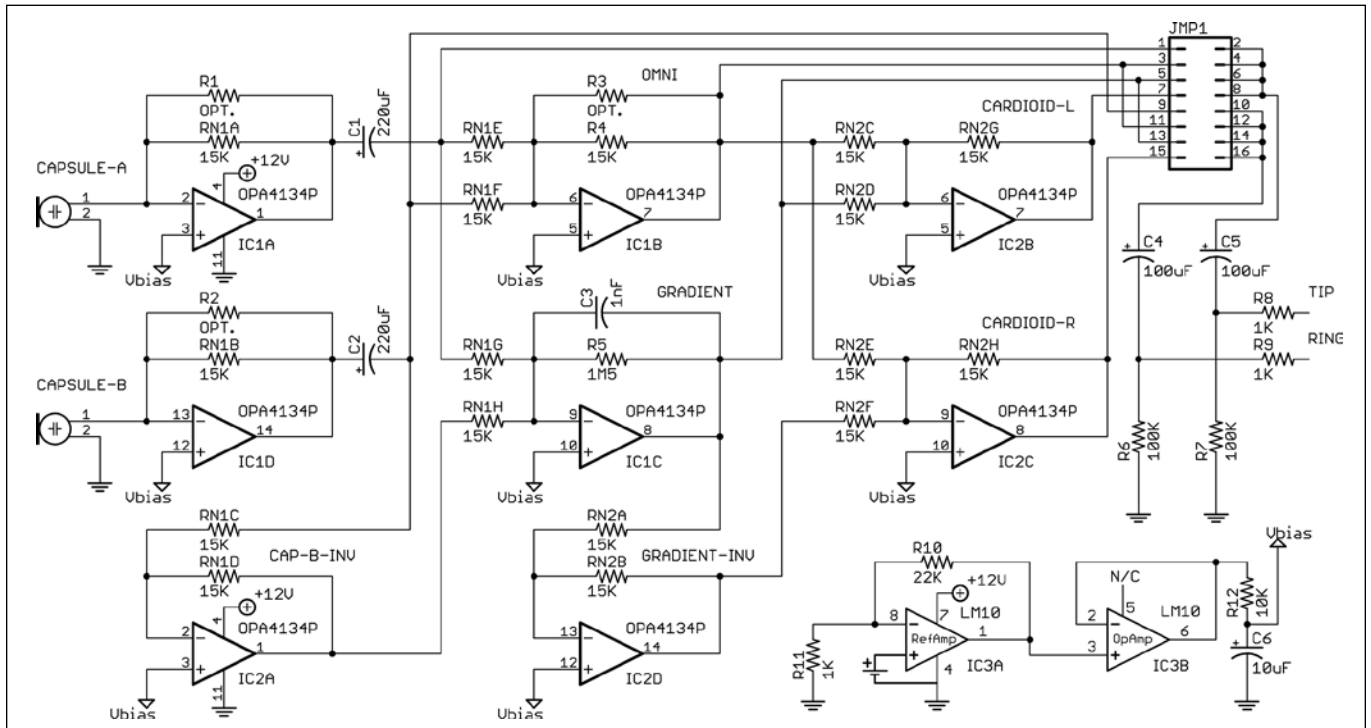
**PHOTO 3:** Analog computer is assembled onto a piece of perfboard with 0.1" hole grid. Power supply and ground bus wires are at top and bottom. Circuit adjustments are made by soldering in optional resistors. Jumper block at right allows various signals to be monitored.



**FIGURE 4:** Simple sine tone burst shown above has discontinuous slope at the start and end. Raised cosine has continuous slope, but gives a DC bump during each tone burst. Adding in the second harmonic with equal amplitude and opposite polarity corrects the DC bump, and gives a continuous band-limited function that can be repeated as necessary to fill a tone burst. Matched filter removes the harmonic during analysis.



**FIGURE 5:** Block diagram of dipole microphone with analog computer. Gradient and cardioid signals are derived using sum, difference, and integrator blocks<sup>1</sup>. Gain stage for each capsule converts current to voltage.



**FIGURE 6:** Detailed schematic diagram for dipole microphone and analog computer. Electret capsules are operated as current sinks at constant voltage. Jumper block JMP1 lets you assign the two output pins to a variety of internal signals.  $V_{bias}$  is set to 4.5V DC, to allow single-supply operation. Required filter and bypass capacitors are not shown.

configure the outputs between measurements.

Instead of providing trimpots for balancing gains, I decided to mount fixed resistors in parallel to accomplish the same thing. I used packaged resistor networks RN1 and RN2 for the 15kΩ resistors, because their values should track well over time and temperature. Coupling capacitors C1, C2, C4, and C5 in the circuit path are all sized to put the low-frequency cutoff well below the lowest frequency of interest. The integrator capacitor C3 is sized to roughly match the gradient and omni signal levels. R5 limits the DC gain of the integrator, and can be used to adjust low-frequency response of the gradient signal, and thus tune the proximity effect in the cardioid outputs.

I used two Burr-Brown OPA4134 quad op amp packages at IC1 and IC2 for the analog computer, mainly so I could become familiar with their characteristics. I found them to have low noise, moderate current consumption, and they are very fast. Layout and decoupling are important, as is an oscilloscope that can see oscillations well beyond 20MHz. Readers may prefer to use TL074 or other unity-gain-compensated op amps in this application. I chose an LM10CN integrated op amp

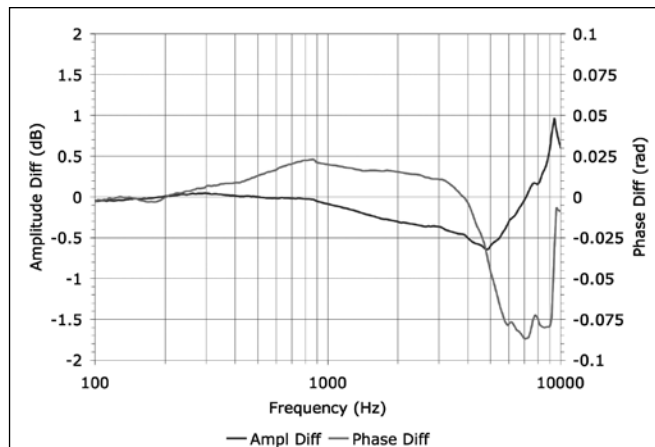
and voltage reference for IC3 to set the bias voltage for single-ended power supply operation.

Instead of just buying two electret microphone capsules and hoping they would match, I actually bought ten capsules (they're not expensive), measured the on-axis phase and frequency response of each, and then chose the two that had the best phase match at 100Hz, while maintaining a similar frequency response all the way up to 10kHz. I actually measured the capsules in pairs, so I could look at their relative phase response instead of the absolute response, and thus subtract out the absolute distance from speaker to microphone. The assumption here is that

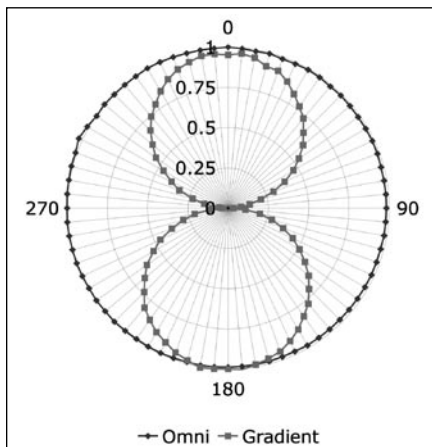
I've precisely set the two capsules equidistant from the loudspeaker. Note that at 100Hz, 1 milli-radian (mrad) of phase difference is just 0.55mm!

The next step is to match the sensitivity of the two microphone cartridges over as broad a range of frequencies as possible. I did this by first measuring the amplitude difference versus frequency for the two capsules when connected to the preamps as shown in Fig. 7. I then calculated the gain change required to match the outputs at 1kHz and soldered in the appropriate resistors at optional locations R1 and R2. The gain and phase differences between the two capsules after this adjustment are shown in Fig. 7, measured using the harmonically shaped tone burst technique.

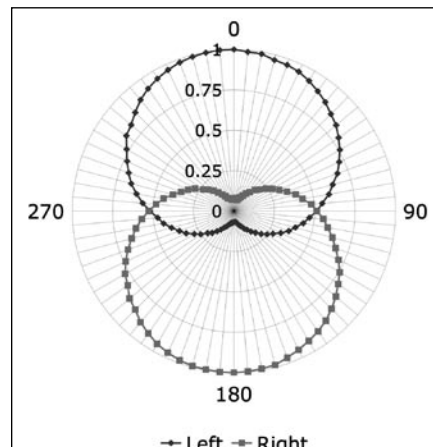
You can now measure the omni and gradient responses, both in frequency and polar domain. The results look encouraging (Fig. 8) with almost 30dB of side rejection in the gradient response at 300Hz. It seems that the integrator design was correct, because the omni and gradient on-axis frequency responses are a good match. To obtain the correct cardioid response, you need to precisely match sensitivity of the omni and gradient responses for sounds coming from the rear, the direction of greatest rejection in the cardioid pattern.



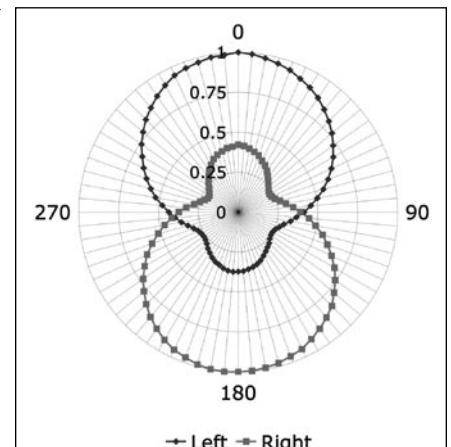
**FIGURE 7:** Measured gain and phase difference between capsule A and capsule B in the dipole microphone prototype. Best match is obtained at 100Hz where it matters most, due to integration of the gradient signal. Note the expanded vertical scale for both amplitude and phase.



**FIGURE 8A:** Measured polar response of prototype dipole microphone, showing omni and gradient signals at 300Hz, 1m distance, with averaging equal to 4. Side rejection is almost 30dB. Plot is normalized to on-axis omni response.



**FIGURE 8B:** Measured polar response of left and right cardioid signals at 300Hz, 1m distance, averaging of 4. Rear rejection is greater than 20dB in both directions. Proximity effect is minimal at this distance. Plot normalized to on-axis response.



**FIGURE 8C:** Measured polar response of left and right cardioid signals at 300Hz, 10cm distance, averaging of 4. Proximity effect causes a loss of rear rejection at this frequency. Plot normalized to on-axis response.

It turns out that the gradient polar response can be asymmetrical due to a small mismatch in phase between capsules A and B, so you may need to adjust the “left” and “right” cardioid balance separately. Gross adjustments can be made by changing the value of integrating capacitor C3. Fine adjustments can then be made by adding resistors at optional R3, or in series with R4 to optimize the left cardioid pattern. The goal is to obtain the best null for rear rejection over the widest range of frequencies.

If your gradient polar response is symmetrical, then the right cardioid response should also be optimum at this point. In my case, however, the gradient response was about 1dB different from front to back, so I adjusted the gradient signal in the right cardioid by adding some resistance in parallel with RN2A or RN2B around amplifier IC2D. Now I have two cardioid responses, one facing front and the other facing back, with rear rejection of 20dB or better in the range from 200Hz to 10kHz (Fig. 9).

You may be wondering whether the sum and difference operations implemented in the analog computer discard any information. It turns out that they don't, in the sense that given the left and right cardioid signals, you can recover the original omni and gradient signals by applying another matrix operation which is the inverse of the sum and difference operation. Expressed as a matrix, it turns out that the sum and difference operator is self-inverse, times a factor of 2.

$$\begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} \begin{bmatrix} 1 & 1 \\ 1 & -1 \end{bmatrix} = \begin{bmatrix} 2 & 0 \\ 0 & 2 \end{bmatrix} \quad (5)$$

This same matrix operator appears in other places in audio, for instance, in stereo modulation of broadcast signals, and in mid-side stereo microphones.

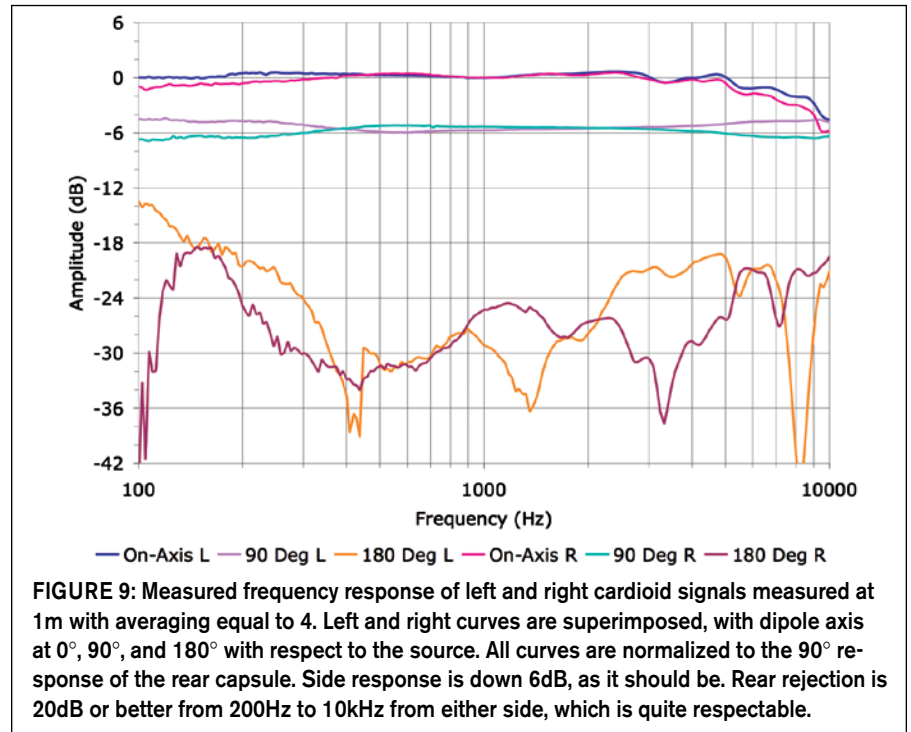
At this point, I've barely scratched the surface of what can be done with directional microphones. I've assumed flat response everywhere, except for rolling off the integrator on the low end. Obviously, much more work can be done with equalization and phase matching here. You could add more capsules to obtain higher order polar patterns or to simulate larger diaphragm microphone designs. The electret capsules weren't modified

for improved headroom, as reported by some authors. I haven't used any acoustic elements in this design, but you could certainly do that as well.

## CONCLUSION

So what does a dipole microphone sound

like in actual use? To find out, I recorded a rehearsal of some musician friends. The left- and right-facing cardioid patterns cleanly divided the group into two bunches, with a few stragglers in between. The bidirectional cardioid pattern could be useful in cases where the only place



**FIGURE 9:** Measured frequency response of left and right cardioid signals measured at 1m with averaging equal to 4. Left and right curves are superimposed, with dipole axis at 0°, 90°, and 180° with respect to the source. All curves are normalized to the 90° response of the rear capsule. Side response is down 6dB, as it should be. Rear rejection is 20dB or better from 200Hz to 10kHz from either side, which is quite respectable.

**TABLE 1 Dipole Microphone Prototype Parts List**

Designator	Part Number	Description	Manufacturer
		0.1µF 50V ceramic disk bypass capacitors	any
		suitable project box	
4 pcs.		#6-32 3/4" metal standoffs	
2 pcs.	137	7/16" dia. brass tubing	K&S Engrng.
	169P44	0.1" grid perfboard	Vector
	AD11B	5/8"-27 mike stand flange	Atlas Sound
	T42	0.042" perf-clip terminals	Vector
2 pcs.		14-pin DIP socket	any
		8-pin DIP socket	any
C1,C2		220µF 35V electrolytic capacitor	any
C3		1nF mylar film capacitor	any
C4,C5		100µF 35V electrolytic capacitor	any
C6		10µF 16V tantalum capacitor	any
Capsule A,			
Capsule B	WM-034BY	Omnidirectional Electret Condenser Microphone Cartridge	Panasonic
IC1,IC2	OPA4134PA	High-performance quad op amp	Burr-Brown
IC3	LM10CN	Op amp and voltage reference	National Semi
JMP1		8-position 2-row header	any
line out		3.5mm 3-circuit phone jack	any
power in		2.5mm 2-circuit phone jack	any
R1,R2,R3		Optional gain trim resistors	any
R4		15kΩ 1/4W resistor	any
R5		1.5MΩ 1/4W resistor	any
R6,R7		100kΩ 1/4W resistor	any
R8,R9,R11		1kΩ 1/4W resistor	any
R10		22kΩ 1/4W resistor	any
R12		10kΩ 1/4W resistor	any
RN1,RN2	761-3-R15K	15kΩ DIP resistor network	CTS
Substitutions are encouraged, in the spirit of promoting research and development.			

you can put a microphone in the middle of the stage. Note that the sum of the left and right signals is just the omni response. This really is a special-purpose microphone as designed, but it holds out lots of promise for future development.

Mechanical noise, coupled from the floor through the mike stand, is tolerable but needs improvement. Signal amplitude is awkwardly positioned between microphone level and line level. Some of the outputs are inverted in terms of absolute phase. Self-noise out of the analog computer is acceptable but needs work. Wind and pop noise sensitivity are high as expected, and readily improved with a generic open-cell foam windscreen. Proximity effect is less than I've experienced with other microphones, and may take some getting used to. Now I can begin the real work of correlating measured results with listening performance, for both purchased microphones, and those of my own design. For more on the test methods used here, go to [www.audioXpress.com](http://www.audioXpress.com).

## REFERENCES

1. A.D. Blumlein, "Sound-Transmission, Sound-Recording, and Sound-Reproducing System," U.S. Patent #2,093,540 (Sept. 1937).
2. J. Eargle, *The Microphone Book*, Second Edition (Focal Press, Oxford, 2004).
3. G.W. Elko and A.N. Pong, "A Steerable and Variable First-Order Differential Microphone Array," 1997 IEEE International Conference on Acoustics, Speech, and Signal Processing, Vol. 1, pp. 223-226 (1997).
4. L.L. Beranek, *Acoustics* (Acoustical Society of America, Woodbury, 1993).
5. H.F. Olson, *Acoustical Engineering* (Professional Audio Journals, Philadelphia, 1991).
6. G.B. Arfken and H.J. Weber, *Mathematical Methods for Physicists*, Fifth Edition (Academic Press, San Diego, 2001).
7. Henning Møller and Carsten Thomsen, "Electroacoustic free-field measurements in ordinary rooms—using gating techniques," Brüel & Kjær Application Note 17-196 (1975).
8. Roger Sanders, "A Simple Shock Mount for Your Microphone," *The Audio Amateur*, #1, 1972, p. 1.

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