WHITE PAPER

HSS®

HYPERSONIC™ SOUND

Theory, History, and the Advancement of Parametric Loudspeakers

A TECHNOLOGY OVERVIEW

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American Technology™ CORPORATION

Contents

The Beginnings of HSS at American Technology Corp

Early Experiments

Without any knowledge of other efforts with parametric loudspeakers, initial attempts at demodulating sound directly in the air at American Technology Corporation were conducted by Elwood (Woody) Norris in his Poway, California garage during the spring of 1996. The experiments used two Goldstar FG-2002C signal generators and two Murata MA40A ultrasonic piezoelectric bimorph transducers. Each transducer was driven independently by a generator in an attempt to hear a non-linearly generated difference frequency. Since the emitters were most effective at around 39kHz, one was driven at 39kHz and the other at 39kHz + xkHz, where x is an audible frequency. If x =1, the difference frequency is 1kHz.

A few transducer configurations were tried, including pairing them side by side (facing the same way), facing them at 90 degrees from each other, and facing them toward each other. The difference frequency effect was noticeable, but very weak (barely audible). The difference frequency could be more easily heard if the second generator's signal was repeatedly swept through a fixed range of frequencies. A range of 40kHz to 49kHz would give a sweeping 1kHz to 10kHz difference frequency. The best results were obtained by pointing the transducers at one another, while placing a tube between them in order to amplify the weak difference frequency signals.

It was thought at the time that the effect could be related to "beats" since the beat frequency is always the difference frequency. Beats are a linear phenomenon, though. When two primary tones are summed or combined, they alternately reinforce and annihilate each other (at the difference frequency rate). The resulting signal (which has a frequency of [f1+f2]/2) will come and go at a slow rate. The ear can only detect beat frequencies of a few Hz. Signals with a higher beat frequency just sound like a continuous tone, with a slight warble. Also, it is important to note that a beat frequency is not an audio signal itself. It is simply the rate at which two higher frequencies go in and out of phase. This is an example of linear superposition.

Woody was able to show early on that the HSS phenomenon was not related to beats. He used a wideband microphone and a spectrum analyzer to show that a sum frequency was also present. It was, of course inaudible, being in the 80kHz range. The exact cause of the tone was a mystery at the time. Woody suspected that there was a "non-linearity" in the air. What it was, we didn't know.

In July of 1996, Woody Norris realized that sending both frequencies through one transducer would ensure that the airborne signals would align and mix properly, and the effect would be maximized because both signals (actually, a single composite signal) travel down a common axis. He rushed to his garage and tried it. He simply hooked both signal generators to one emitter. The effect could be clearly heard now. Soon, more transducers were added to make clusters for more output. They were wired together so that the overall output was the sum of each individual emitter. The transducers were arranged in a hexagonal pattern to maximize the number of emitters in a given space. Piezo bimorphs from both Murata and Nicera were used, as were the lower output Polaroid electrostatics. As previously stated, at ATC it wasn't known at the time that this had been done before in other parts of the world. Still without the benefit of knowledge of the prior art, the first HSS patent filings were dated July 17, 1996.

ATC's Discovery of Parametric History and Theory

Non-Linear Acoustics 101

A search was made at the San Diego State University library for references related to sum and difference tones. The Journal of the Acoustical Society of America had several papers mentioning sum and difference tones, but none found gave further theory or information. The phrase "Parametric Array" was still unknown at ATC, so searches were not performed under that name.

An acoustics text with a translated paper by Helmholtz was found. Titled, "**On Combination Tones**", this paper provides a theory that is not currently accepted in the non-linear acoustics arena, but it is worth a brief look. In this paper, Helmholtz introduces his novel theory on how combination tones (sum & difference) are generated by a non-linear restoring force on a displaced molecule. These were not subjective tones (a product of psychoacoustics, as were so-called "Tartini Tones"), but ones that actually existed in the air.

He surmised that the 'springs' that keep air molecules spaced apart exhibit a non-linear restoring force characteristic that manifests itself at higher displacement amplitudes.

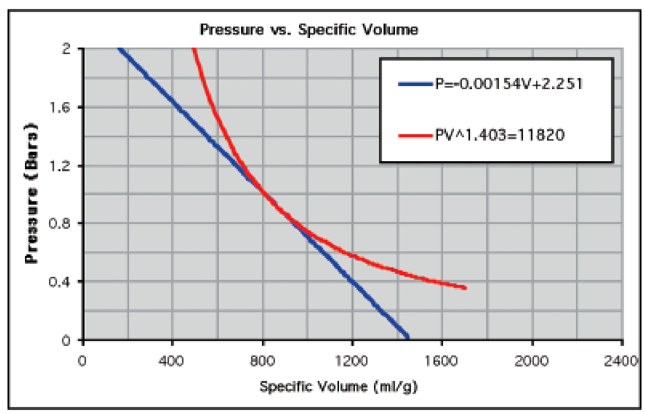
Helmholtz's theory and formulas predicted results that initially seem to match what Helmholtz had measured (and what we measured), so it would seem to be a valid theory.

However, the difference frequency amplitude term in his formula is interesting in that it predicts a rising level with a decreasing difference frequency which did not match our experimental results. Other than two special cases we'll return to later, all HSS experiments to date had shown a falling level for decreasing difference frequencies. Therefore, this theory needed to be set aside and a new one found.

Since two primary frequencies (in our case, ultrasonic ones) are generating new frequencies in the air, the shape of the primary wave must change as it propagates. Fourier tells us that any wave can be described with a series of sines and cosines. If one emitts two high-amplitude sine waves, as in the Helmholtz example above, new frequency terms appear, and the shape of the wavetrain changes.

The accepted mechanism for this "propagation distortion" is explained by A.L. Thuras, R.T. Jenkins, and H.T. O'Neil of Bell Labs in a 1934 paper called **Extraneous Frequencies Generated in Air Carrying Intense Sound Waves,** *J. Acoust. Soc. Am.* 6:173-180 (1935). The following explanation is taken from this paper, and from a similar paper by L.J. Black, **A Physical Analysis of Distortion Produced by the Non-Linearity of the Medium,** *J. Acoust. Soc. Am.* 12:266 (1940).

It turns out that if equal positive and negative increments of pressure are impressed on a mass of air, the changes in the volume of the mass will not be equal. The volume change for the positive pressure will be less than the volume change for the equal negative pressure. This can be seen in a plot of pressure vs. specific volume (1/r).



This is a qualitative picture of the relationship between pressure and the inverse of density (the so-called specific volume).

This phenomena may be unfamiliar to those in the relatively linear acoustics field of audio. The wave equation which is customarily used in the solution to acoustical problems is valid for small signal propagation only. The assumption involved in the derivation of the small signal wave equation is that the maximum displacement of the air particles x be small compared to the wavelength l; x < l. In other words, the pressure fluctuations are so small that the specific volume appears to be a linear function of pressure. When this is not satisfied, a plane wave or even a spherical wave propagated in the medium will not preserve its shape. As a result, the magnitude of the fundamental decreases and the magnitude of the distortion increases with propagation distance. A simple explanation of this phenomenon is given by L.J. Black. It goes as follows:

Each part of the wave travels with a velocity that is the sum of the small signal velocity and the particle velocity.

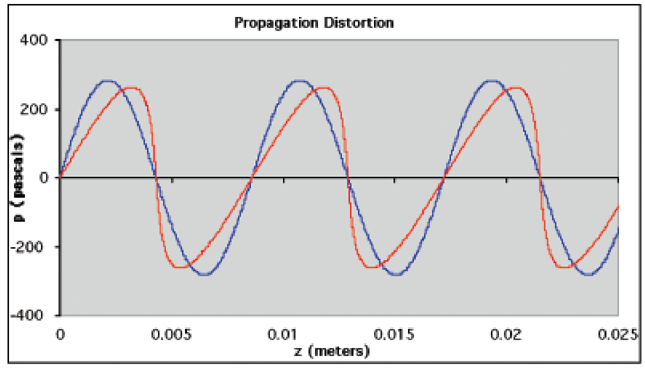
The maximum condensation in a wave is at the point of maximum pressure and this portion of the wave has the greatest phase velocity. The fact that the phase velocity is greater at the peak of the wave than at the trough results in a wave whose shape changes continuously as it is propagated.

Assuming the normal small signal velocity of c = 344m/s and working through the equations with a sound pressure level of 140dB (re. 20µ Pa) signal, gives a result of c ≈ 344.8m/s at the pressure peak of a wave.

This figure is only 0.24% higher than the small signal speed of sound. However, it says that the peak will travel 80cm further than the ambient part of the wave in only one second. That's a lot considering the wavelength for a 40kHz signal is only 8.6mm. In a 40kHz sinewave, the peak begins propagating only 2.15mm behind the ambient (zero crossing), so for a 140dB signal it only takes 0.0027s, or about 1 meter of propagation for the peak to catch up to the zero crossing.

It should be noted that along with a traveling pressure wave, there are associated density and temperature waves too. In other words, the density and temperature at a point in space also varies with time at the same frequency as the pressure wave.

The following chart shows the shape of a wave which has been distorted by the mechanism explained above. The blue line represents a pure sinewave (a single-frequency signal); the red line represents the shape of the same wave after it has propagated through the non-linear medium for a time.



The values used for the graph above are as follows:

SPL of the pure sinewave: 140dB (re. 20µ Pa)

Frequency: 40.0kHz

Propagation time: 0.0013 sec.

It can be seen that a high-amplitude sinewave tends to form into a sawtooth wave as it travels. The sawtooth wave contains odd and even harmonics. The 2nd harmonic is fully half the amplitude of the fundamental. This means that strong harmonics are created during the propagation of a high-intensity tone. Things get more interesting when there is more than one primary, or fundamental, tone. In the two-tone case, f1 & f2, it can be shown that the harmonics of each will appear, as will the sum and difference frequencies, f1+f2 and |f1-f2|. This is the most simple case of a parametric acoustic array.

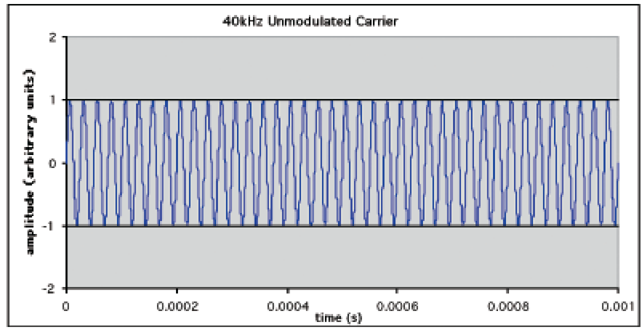
The History of Parametric Acoustic Arrays in Air

In late December of 1962, a Physics Professor from Brown University, Peter Westervelt, submitted a paper called **Parametric Acoustic Array,** *J. Acoust. Soc. Am.* 35 (4):535-537 (1963). Westervelt considered primary waves interacting within a given volume and calculated the scattered pressure field due to the non-linearities within a small portion of this common volume in the medium. He goes on to describe the generation of a difference frequency tone along the collimated beam of a two-frequency signal. Many simplifying assumptions were made, such as: no attenuation of the difference frequency, a perfectly collimated primary signal beam, the two primary signals attenuate at exactly the same rate, etc.

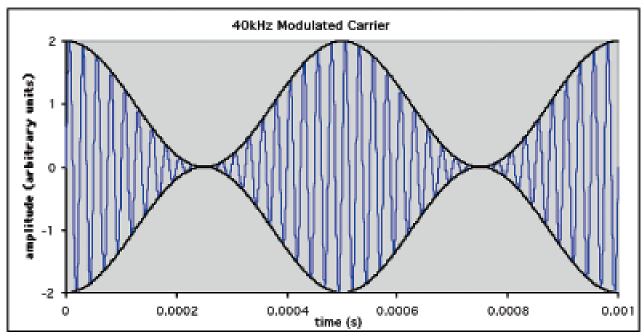
His was not as complete as the currently accepted theory, but it was close. Some further refinement was forthcoming, though.

In 1965, H.O. Berktay published the first paper that gave an accurate and more complete theoretical explanation of the parametric acoustic array: **Possible Exploitation of Non-Linear Acoustics in Underwater Transmitting Applications,** *J. Sound Vib. 2* (4):435-461 (1965). His analysis was more general and more complete. He covers cases where the primary signals are expanding cylindrically or spherically, as well as the collimated plane wave case we are interested in. He also didn't limit the analysis to the two-tone primary case. Rather, he uses the concept of the modulation "envelope". This is very powerful because a parametric loudspeaker isn't usually limited to making one tone at a time. The envelope analysis allows us to look at any primary signal spectrum. It turns out that the demodulated signals (the ones we are ultimately interested in) depend on this envelope function.

To illustrate the concept of an envelope, it's useful to look at some examples. An unmodulated 40kHz carrier signal looks like this in the time domain (what you would see on an oscilloscope):



The black line running across the peaks of the wave is the envelope. Notice there is also a black line running across the troughs. This can also be used as the envelope boundary. As a rule, the lower envelope is a mirror image of the upper one. Below is the same 40kHz carrier, now being amplitude modulated by a 2kHz tone.



The envelope function in this case is $E(t)=1+\cos(2*\pi*2kHz*t)$, or $-[1+\cos(2*\pi*2kHz*t)]$. With an amplitude modulated carrier, the envelope is simply proportional to the modulation signal, plus an offset constant. Also notice that the lower envelope is a mirror image of the upper one.

Berktay assumes the primary wave has the form

$$P_1(t) = P_1E(t)sin(\omega_c t)$$

where ω_c is the carrier frequency and $E(t)$ is the arbitrary envelope function. For a point along the transducer radiation axis, the secondary (or demodulated) signal is

$$p_2(t) = \frac{P_1^2 A}{16\pi\rho_0 c_0^4 z\alpha}\frac{\partial\beta}{\partial t^2}E^2(\tau) \text{ where}$$

$$\tau = t - \frac{z}{c_0} \text{ is the lag time, and}$$

$$\beta = \frac{+1\gamma}{2}$$

A is the cross-sectional area of the primary beam, z is the distance along the beam, andα is the attenuation factor of the primary signal. The amount of demodulation decreases with distance, but the demodulated signal at a given distance sums in phase with what has already been created. Therefore, a virtual end fired-array is realized. Near the emitter, the demodulated signal level actually increases with distance. It should be noted here that the effective array length is $1/\alpha$, where a is the amplitude absorption coefficient.

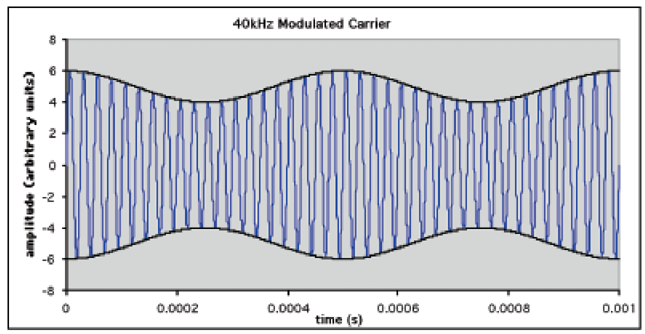
To summarize the Berktay solution: the demodulated signal in a parametric array is proportional to the second time-derivative of the envelope squared:

$$p_2(t)\alpha\frac{\partial^2}{\partial t^2}E^2(t)$$

This is called "Berktay's Far-Field Solution" because he assumes we are far enough away from the source to ignore the presence of any ultrasound (or primary frequencies). The solution is valid for the near-field, but the ultrasonic primaries co-exist with the demodulated signals. This is the fundamental expression for the output of parametric acoustic arrays.

If we ignore the linear frequency response for the moment and use the Berktay solution to derive the non-linear result of inputting a single 2kHz tone into a double sideband amplitude modulator we get an output that shows that two pure tones result from this envelope, and they both have the same amplitude. The 2nd harmonic, 4kHz, has the same amplitude as the 2kHz fundamental. This is equivalent to 100% THD. It is clear that raw AM (amplitude modulation with a carrier) does not give the desired results. The envelope must either change its frequency content or its offset.

First, lets maintain the 2kHz envelope, but increase its offset. This gives a time-domain signal that looks like this:



To get this signal, the carrier amplitude in the figure shown above, was simply increased by a factor of five. This results in a decreased modulation percent. The earlier case was 100% modulated, where this one is only at 20%.

Now the desired 2kHz fundamental frequency dominates. It is 10.5 times greater in amplitude than the 2nd harmonic. This is ≈ 19% THD. The distortion has been reduced by a factor of five, but the carrier level had to be increased by the same factor. The audio level has also increased by a factor of 5.25. Had the modulation % been maintained while the power was increased by a factor of five, the audio level would have increased by a factor of 25. By reducing the modulation index we are giving up conversion efficiency for a cleaner signal. While the THD is reduced, it still isn't very good and our efficiency is poor with this method. More on this quandary later. This is just a preliminary look at Berktay's solution.

Some years went by and Westervelt made a presentation on non-linear acoustics applications. He claimed that the array would not work in the air. We'll let Dr. Blackstock (University of Texas) pick up the story from here:

> *"The exchange occurred in 1971, at a meeting sponsored by the Navy in Washington, DC. My recollection is that the meeting closed with informal presentations by one or more participants. One of these was Peter Westervelt. He observed that no one up to now had reported a successful parametric-array-in-air experiment; moreover, there had been rumors of a failed classified experiment on the subject. He then proceeded to give theoretical reasons why the parametric array should not be expected work in air. Orhan Berktay and I were sitting together, and we exchanged very skeptical glances. When Peter sat down, very close to where we were sitting, I couldn't contain myself. I had the audacity to say, 'Bullshit, Peter.' I've often wondered since then how I had the guts to say that to the Grand Old Man of parametric arrays. Peter retorted 'All right, you prove it!'*
>
> *"The gauntlet having been thrown down, after returning from the meeting, I contacted Mary Beth Bennett and suggested that she do an experiment to demonstrate that the parametric array does indeed work in air. I'm a bit hazy about the events at this point, but I believe I consulted Elmer Hixson, who had been an advisor of Mary Beth's, before I spoke with her. He thought it was a good idea but warned that it might be a tough experiment to do (he was right). I guess you could say that Mary Beth pulled my chestnuts out of the fire."*
>
> *— David Blackstock*

Soon, Mary Beth Bennett was busy with the Parametric Array in Air experiments. In 1974, Mary Beth and Dr. Blackstock presented their results. **Parametric Array in Air**, *J. Acoust. Soc. Am.* 57 (3):562-568 (1975). They showed once and for all that the parametric array can be realized in the air.

They used an oil-filled hydrophone with output modes at 18.6kHz and at 23.6kHz (for a difference frequency of 5kHz). The emitter can be seen today in Mary Beth's office. It has a circular center section for one frequency mode and a second concentric ring section for the other mode. Unfortunately, they had to deal with spherically spreading primary signals, and poor energy coupling to the air. Even so, they were still able to show that the parametric array did indeed exist in the air. They even showed an increasing amplitude difference frequency signal when the propagation distance was increased.

Prior Art Efforts

While working for another company, in late 1996, Jim Croft, now Sr. Vice President of Research & Development at ATC, provided his comprehensive file of parametric loudspeaker historical papers and over the next few months completed our parametric library with a group of translated papers on parametric theory and development.

It had now been brought to ATC's attention that other individuals and companies had previously taken an interest in the parametric array in air. It turned out that many research papers have been published on the subject. Most of the efforts directed at creating a practical device were based in Japan. The most notable authors of these works are Yoneyama, Kamakura, Kumamoto, Aoki, and Ikegaya.

In a paper called **The Audio Spotlight: An application of nonlinear interaction of sound waves to a new type of loudspeaker design**, *J. Acoust. Soc. Am.* 73 (5), May 1983: pp1532-1536, Yoneyama, et al, makes an attempt at reasonable performance with off-the-shelf components. The authors built a large hexagonal array of PZT bimorph emitters (547 pieces) that operate at around 40kHz, with a secondary peak just below 60kHz (typical of this type of emitter). In order to overcome the 12dB/octave frequency response slope predicted by Berktay, they propose that they should equalize their incoming audio signal before modulating the carrier with double sideband AM.

This gives the desired frequency response (over a limited range) by decreasing the modulation index for higher audio frequencies. For a double sideband AM case with one audio frequency input, the demodulated audio signal amplitude will be proportional to m, the modulation index. The second harmonic amplitude is proportional to m2. In order to achieve low distortion with DSB AM, m must be made as small as possible, at the expense of efficiency. This is done to minimize "cross interaction" between the upper and lower sidebands, they say. Since 12dB/octave is a severe slope and the audio band is wide, a high modulation index must be used for the lower audio frequencies. A high amount of distortion would result for these frequencies.

In the actual experiment, they simply use the peak of the emitters frequency response to equalize the audio signal. They achieve a flat response from about 1.5kHz to about 7kHz. They also show sharp directivity for all audio frequencies. However, their experiments show that the directivity decreases for lower audio frequencies.

At the end of the paper, they state that such a system could be used in a museum without sound barriers between exhibits.

The next paper was from 1984, and it was presented at the 10th International Symposium on Nonlinear Acoustics. It's called **Developments of Parametric Loudspeaker for Practical Use** (1984). They attack three problems. The optimum carrier frequency, distortion correction, and insulating the listener from the ultrasound.

They state that the carrier frequency needs to be as low as possible, but not so low as to cause beam spreading. They recommend a frequency between 30kHz to 70kHz. In order to correct for distortion, they introduce the concept of square-rooting the modulation signal which makes a lot of sense based on the Berktay far field solution stating that the envelope is squared by the air. They quickly discover that this will require additional bandwidth. A problem we'll explore more fully later on.

Their experimental device in this work consists of 581 PZT bimorphs arranged in a square package. They were able to show that for a few input cases, the second harmonic output can be greatly reduced while using square rooting. Finally, to minimize ultrasonic output, they propose an acoustic filter to attenuate the ultrasound at some arbitrary distance from the emitter.

The next paper is called **A Study for the realization of a parametric loudspeaker**, *J. Acoust. Soc. Japan*, June 1985. Again, they look at a few problems. First, the low conversion rate (from electrical to audio). They revisit distortion correction and safety again too.

In order to improve efficiency, they suggest a lower carrier frequency to make the array longer. Towards the end of the paper, they suggest 30kHz as a good trade-off between beam length, spreading, and safety. They also reiterate that a higher ultrasonic SPL level increases gain, not just output, but they do not account for saturation effects. They do discuss how increasing the aperture size can be traded for a lower primary SPL. This is very important.

On the distortion front, they speak in terms of a single frequency audio signal. There are two ways to clean this up. One is square rooting, which results in many sidebands. The other is by using two discreet primary tones (which is SSB) so the "interaction of two sidebands" is eliminated. They don't realize that the envelope in each case is the same, they only talk about removing the sideband interaction. For more complex signals, they admit that SSB distortion results from "non-linear interaction of the sideband itself". This means that for complex signals, only square rooting with DSB will work, but with the bandwidth penalty.

Finally, they use some packing material (a 4 wide by 5 feet long piece) to absorb the ultrasound before the instrumentation (the mic). Unfortunately, this reduces the audible sound levels too.

The next work they published appeared in Acustica: **Suitable Modulation of the Carrier Ultrasound for a Parametric Loudspeaker,** *Acustica*, Vol. 73, 1991, pp215-217. This is a short paper, and it deals with reducing the radiation power requirement. Specifically, they discuss reducing the carrier level when there is no sound or when there is little volume needed. We call this a dynamic carrier. Where the envelope will be flat, the carrier is suppressed altogether. This can save a lot of power, but does introduce additional nonlinearity.

For the experimental portion, they use an even larger array of 2000 bimorphs resonating at 28kHz. They used a speech for the tests (84 seconds of a male news announcer). It is claimed, that without degradation of the demodulated signal, they were able to reduce the power requirements by more than 64% (390 Watts average power to 140 Watts average power for the dynamic carrier version). Impressive results.

Next is their paper called **Parametric Loudspeaker—Characteristics of Acoustic Field and Suitable Modulation of Carrier Ultrasound,** *Electronics and Communications in Japan*, Part 3, Vol. 74, No. 9, 1991, pp76-81. There isn't much new ground broken here. They use a large array (1.4ft dia.) of 1410 emitters working at 27kHz and 30kHz to create a 3kHz audio tone.

The main item of note in this work is an assertion (based on Merklinger's earlier work, **Improved Efficiency in the Parametric Transmitting Array,** *J. Acoust. Soc. Am.* 58 pp. 784-787 (1975)) that for smaller signal levels, the demodulated signal is proportional to the envelope squared (Berktay). With high amplitudes, they state that the audio signal becomes proportional to the envelope itself. They go so far as to say that this occurs around 130dB.

This would mean that square-rooting requirements for distortion correction would go away with primary signal levels over 130dB. We have tried to verify this in our lab with no success. The single tone SSB signal sounds cleaner at all signal levels, even above 140dB. Also, in response to our inquiry, Dr. Blackstock has given this some thought and he isn't willing to say that they are correct.

Next up is a paper called **Parametric Sound Radiation from a Rectangular Aperture Source,** *Acustica*, Vol. 80, 1994, pp332-338. The main point of the paper is to compare the results of their experiment using a 24cm x 44cm (9.4in x 17.3in) device generating 25kHz and 30kHz, with the KZK (Khok-hlov-Zabolotskaya-Kuznetsov) non-linear parabolic equation. The KZK equation combines non-linearity, dissipation, and diffraction in beams. It is a successive approximation method. It covers the generation of harmonics from single-tone primaries. The KZK equation has been shown to have good agreement with experimental results when circular apertures were used, but the authors wanted to try a rectangular source. They concluded that in the far-field, the rectangular source gave roughly the same results as a circular source of equal area would have.

For the experiment, they used 1102 small piezo bimorphs. They were driven via a two-frequency signal as mentioned above. The source pressure was only 116dB, so little difference tone level is expected. However, they did get a 79dB 5kHz signal at 3.5m (which was the best distance for this setup). When they went to a 128dB source level, they got more than 100dB of 5kHz at 3.5m. They do say that the harmonics (distortion) predicted with a rectangular source are lower than those predicted for a circular source. Finally, they complain of the computation time for far-field KZK analysis.

A final paper was published by the Japanese group called **A Parametric Loudspeaker—Applied Examples,** *Electronics and Communications in Japan*, Part 3, Vol. 77, No. 1, 1994, pp64-73. As the title implies, the authors tried out some actual applications to see where the technology can be used successfully. They also do some theoretical work using a transformed beam equation that is solved by “using an implicit backward finite difference scheme based on the Richtmyer method.”

They attempt first to make a compact emitter array (so as to be practical) so they must consider the best possible carrier frequency to maximize the ultrasound to audio conversion. As they did previously, it is stated that lowering the carrier makes for a longer beam, but spreading begins to increase if the carrier is lowered too far. They also make the case that shock formation is a big inhibitor of difference tone generation. This would seem to be intuitive. If the waveform cannot change shape further, it cannot continue to effectively make audio signals.

They compare the audio generated by 25kHz and 30kHz primaries with the audio generated by 50kHz and 55kHz primaries. Not only is the difference tone slightly stronger (in the near field) from the lower frequency primaries, but the harmonic distortion is lower too. In the far field, the difference tone is stronger yet for the low frequency case.

They state that the optimal carrier frequency is about 35kHz, but that higher frequency primaries may be safer to humans. They then construct an array of 91 piezo bimorphs (1cm in dia. each). The overall size was 11cm in diameter, and they drove it with 30Vpp. The two primaries are 38.5 and 41.5kHz for a difference of 3kHz. They estimate a 134.5dB source level, and a 90dB difference level at about 0.5 meters. They then generated bird chirping sounds (similar to a crosswalk signal) and used less than 100dB of ultrasound.

Finally, they compared a larger array (2208 small piezo bimorphs) against a horn loudspeaker of similar aperture in a reverberant tunnel to check for speech articulation. The tunnel was long (1777m) and had 52dB of background noise (78dB when there are idling engines). They also filled part of the tunnel with cars and trucks. They cite a subjective intelligibility scale of which the parametric speaker wins in monosyllable tests and three-syllable tests (the margin in the monosyllable test was close, they blame the modulator). They did say that in the noisy condition, the output was too low. But in the quiet, the parametric speaker communicated better. Our experience agrees with these findings.

As mentioned earlier, most of the prior art has been from Japan, but there has been some work done in other parts of the world, and domestically as well. While there haven’t been any recent papers that break significant new ground regarding parametric arrays, it is worth citing one more paper of note.

It is titled: **The Use of Airborne Ultrasonics for Generating Audible Sound Beams,** *Audio Engineering Society*, Presented at the 105th Convention, 1998 San Francisco, CA, by F. Joseph Pompei of the MIT Media Lab.

In this paper there is some mention of the prior art but, as it was with ATC in the early days, he appears to be unaware of much of the work that had been done previously. The paper goes on to provide a good discussion the challenges of performing the standard preprocessing scheme. That is, double integration and square rooting. It is noted that infinite bandwidth is needed for a perfect implementation of this scheme, consistent with the Japanese view, but it is considered that the sidebands decrease in level as they are further removed from the carrier such that a reasonably wideband ultrasonic device may suffice.

A large array (≈ 35cm) of 60kHz (center frequency) wideband devices is used. Since the frequency response in this paper is a near-perfect match to the Polaroid single-ended electrostatic ranging devices, referred to in early ATC experiments, it is assumed that these are most likely the devices utilized. The main theme is the need for wideband ultrasonic devices to recreate the square-rooted audio DSB spectrum. The distortion reduction achieved with square-root preprocessing is significant and consistent with the expected results based on the earlier Japanese research.

Figure 5 in his report is curious in that it shows higher SPL at 400Hz than it does at 5kHz. This is likely due to experimental error. Using a flat (or nearly so) response emitter results in about 12dB/octave of rolloff in the audio signal. A microphone can be fooled (even a good one like the B&K 4138 or the new 4939). It may be that the radiation pressure of the ultrasonic wavetrain is giving a false reading. The radiation pressure will change at an audio rate when doing AM, and will deflect a microphone diaphragm at lower frequencies showing what appears to be much more extended low frequency bandwidth than what is measured with a properly filtered microphone or perceived audibly.

By the last half of 1997 research at ATC combined with the knowledge gained by the pre-1990 papers gave the R&D group at ATC a knowledge base for parametric arrays that allowed effective movement forward in pursuing and creating novel solutions for the remaining problems inherent in parametric technology.

Parametric Problems Remaining to be Solved in 1997

With the state of the parametric loudspeaker art well defined and further advancements having stalled over ten years ago, it was now clear, through exposure to the body of previous parametric investigation combined with extensive research at ATC, what problems remained and what improvements had to be made in each of the three main system categories if a practical and useful device were to be realized.

Some Significant Problems of Prior Art Systems

1) Double Sideband and Preprocessed Double Sideband Signal Processing

- high distortion or,
- low modulation index causing low conversion efficiency, or
- unrealistically wide bandwidth and high carrier frequency requirements to achieve the ideal modulation envelope which when implemented causes reduced output and/or correction terms appearing in the audible range as another form of very audible, non-harmonically related distortion;
- upper-to-lower sideband asymmetries limiting the effectiveness of preprocessing

2) Piezoelectric Bimorph Transducer Arrays

- mismatched transducers in multi-unit arrays
- sideband amplitude aberrations
- out-of-band subharmonics falling into the audible range
- expensive multi-transducer arrays (500 or more units)
- high distortion
- unreliable for sustained outputs of the required levels
- insufficient ultrasonic bandwidths

3) Linear Ultrasonic Power Conversion

- high dissipation of sustained half voltage carrier operating at lowest efficiency
- very high dissipation due to reactive load transducer interaction
- poor load matching/power transfer

Types of Solutions Required

1) Signal Processing

- A method for modulation and distortion reduction that:
  - **a)** is able to minimize distortion by creating output that matches the ideal modulation envelope while simultaneously;
    - **i)** does not increase bandwidth requirements (preferrably even reduces bandwidth)
    - **ii)** allows high modulation index for good conversion efficiency
    - **iii)** allows the lowest possible ultrasonic operating frequency for greatest output
    - **iv)** eliminate the problem of upper to lower sideband symmetry

2) Emitter Design

- An emitter that removes the limitations of the prior art multiple bimorph transducers by:
  - **a)** eliminating the multi-transducer unit-to-unit variations
  - **b)** exhibiting very high efficiency, particularly at the carrier and low frequency correlated sideband frequencies
  - **c)** having adjustable resonant frequency adaptive to various carrier frequencies & modulator types
  - **d)** elimination of out of band (audio range) subharmonics of the prior art devices
  - **e)** a monolithic structure for matched output over surface and repeatable, simplified construction

f) having ultrasonic bandwidth at least equal to that of the audio source

g) having greater than 140dB large signal capability

h) having inherent low distortion

3) Ultrasonic Power Conversion

- An ultrasonic power converter system that provides:

  a) good efficiency at low crest factors

  b) efficient drive into ultrasonic reactive loads

  c) impedance matching for efficient power conversion into the load

Further it is important to work across the technology areas to:

- integrate a narrow band modulation scheme with a transducer technology such that the combination eliminates any unwanted audioband artifacts
- integrate the power amplification intimately with the emitter load to maximize efficiency and effective energy transfer
- correlate emitter resonant frequency, modulation carrier frequency, and amplifier matching at those frequencies through an integrated rather than independent development

Through extensive research significant advancements have been made in each area of parametric loudspeaker system design, solving the above listed problems and providing further improvements.

A Partial List of Key ATC Proprietary Solutions

1) SIGNAL PROCESSING

- Half bandwidth modulation w/unprocessed zero distortion simple signals
- Zero bandwidth distortion correction
- Psychoacoustically favorable modulation method

2) EMITTER DESIGN

- Monolithic film ultrasonic transducers

  Electrostatic, Piezo-electric Film, and Planar magnetic emitters

  Pressure based PVDF

3) POWER CONVERSION

- High efficiency ultrasonic power amplifiers
- Switching frequency/carrier frequency correlation
- Reactive power regeneration
- Switching filter/transducer impedance matching integration

Signal Processing

In order to convert the source program material to ultrasonic signals, a modulation scheme is required. In addition, error correction is needed if distortion is to be reduced without loss of efficiency. The goal, of course, is to produce the audio in the most efficient manner while maintaining acceptibly low distortion levels.

We know that, for a DSB system, the modulation index can be reduced to decrease distortion, but this comes with a cost of reduced conversion efficiency. The other choice that has been explored by us is so-called “square-rooting” of the audio before modulation. This gives the proper envelope for a DSB system and allows for a high modulation index, but it requires large bandwidths when implemented properly. In fact, since the lower sidebands extend out so far below the carrier, the carrier frequency needs to be pushed much higher in order to keep sidebands out of the audio range. On the upper sideband, ultrasonic absorption, among other impediments, limits the ability to produce the desired bandwidth. But, a high carrier frequency is also less efficient for parametric conversion than a lower one.

The three signal processing performance issues/problems with parametric loudspeakers are: high distortion, low efficiency/output, and an unreasonable bandwidth requirement. Any two of these can be easily overcome, but as a direct result the third problem will be maximized. This quandary is explored in the following paper: **Parametric Array in Air: Distortion Reduction by Preprocessing,** Thomas D. Kite, John T. Post, and Mark F. Hamilton. *Proceedings of the 16th International Congress on Acoustics and the 135th Meeting of the Acoustical Society of America*, 20-26 June 1998 p1091.

They explore the theoretical results of raising and lowering the modulation index, the carrier frequency, and the use of square-rooting. It is assumed in the paper that the only way to get a square-rooted envelope is to use an audio square-rooter before the modulation step. One of the authors (the faculty advisor to this paper), Mark Hamilton, had even said that the *only* way to get a square-rooted envelope was with an audio square-rooter. This is not the case, as will be demonstrated below. Mark Hamilton, David Blackstock, and Elmer Hixson were all shown how it could be done during a presentation by Joe Norris (co-author of this document) while presenting ATC's technology at the University of Texas.

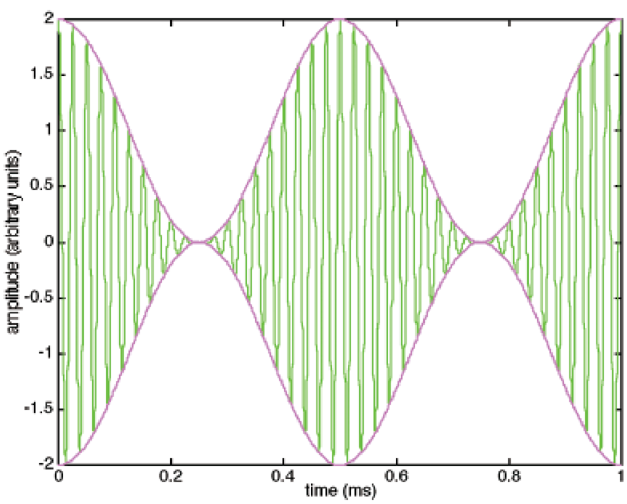
Since there is a direct relationship between the modulation index and the resulting conversion efficiency, a high modulation index is desirable. So how can we achieve low distortion without a high bandwidth requirement? If square-rooting were the only option, ideal distortion reduction of a full band audio signal would be very difficult to achieve. Remember, Berktay's solution says that the audio signal will be proportional to the envelope, not the spectrum. While each spectrum has only one envelope, fortunately, there is more than one spectrum that can be used to achieve the ideal envelope shape. How this can be turned into an advantage will be explored below.

In the prior art parametric loudspeakers, two basic approaches have been utilized; double sideband amplitude modulation and double sideband amplitude modulation with square root preprocessing (always with a carrier when used for parametric conversion).

In terms of general approaches to amplitude modulation for non-parametric uses, there is another scheme called Single Sideband (SSB) amplitude modulation. While previous parametric loudspeaker design has been substantially limited to DSB, SSB should also be analyzed and compared to determine the advantages and disadvantages of each. Since optimal demodulated audio signals depend only on the envelope shape (and not the ultrasonic spectrum), there is substantial freedom in choosing the modulation scheme. Assuming Berktay's Far-Field solution for the nonlinear wave equation is valid, it turns out that a single-sideband amplitude modulation scheme holds some interesting and important advantages over a double-sideband AM scheme.

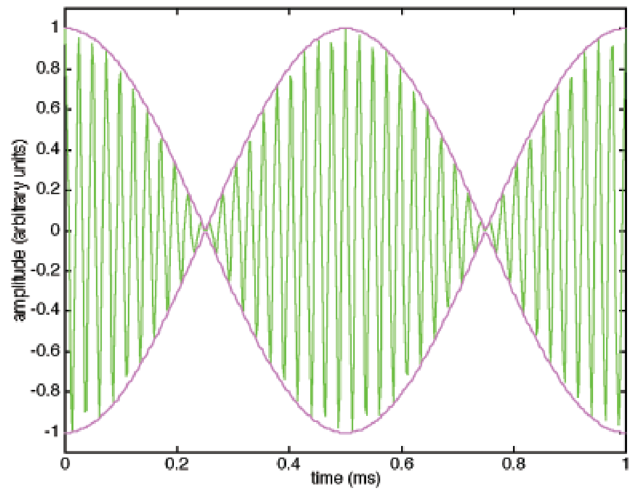
The ideal envelope shape can be determined quite easily. It is simply the square-root of the desired audio output, plus any equalization factors. The goal of any modulation scheme should be to generate a modulation envelope that matches as closely as possible this ideal envelope. It is instructive to compare the modulation schemes and their resulting envelopes with the ideal envelope. For each test case evaluated below, a 40kHz carrier will be used for simplicity. The maximum modulation index will also be used (because the differences between the modulation schemes are less dramatic at low modulation indexes). Note: The two time-derivative operations in Berktay's solution translates to a 12dB/octave high pass slope in the parametric audio output. This will be ignored for now since it can be corrected independent of the modulation scheme.

For DSB, the resulting spectrum is simple, being just two equal-amplitude sidebands at 38kHz and 42kHz, centered around the 40kHz carrier. It will be assumed that the envelope function is proportional to (or the same as) the modulation signal shape. To achieve this, the DC off-set modulation signal is simply multiplied by the carrier. For example, a 2kHz audio signal (used as the modulation signal) would result in a 2kHz sinewave envelope. The following is the time-domain signal of these three frequencies, 38kHz, 40kHz, and 42kHz:



When this envelope is squared by the process in the medium of air, the 2nd harmonic ends up with the same amplitude as the fundamental or 100% THD as was mentioned earlier in the paper.

For SSB, it will be assumed that the incoming audio signals will be frequency “up-shifted” by the carrier value. For example, a 2kHz audio tone will be up-shifted by 40kHz, resulting in an upper-sideband frequency of 42kHz (in addition to the ever-present carrier). For the single audio tone case (40kHz carrier, 2kHz audio input) the resulting spectrum will be simply two equal-amplitude sinewaves at 40kHz and 42kHz. The following plot shows the time-domain signal of these two frequencies in green, and the resulting envelope in magenta.

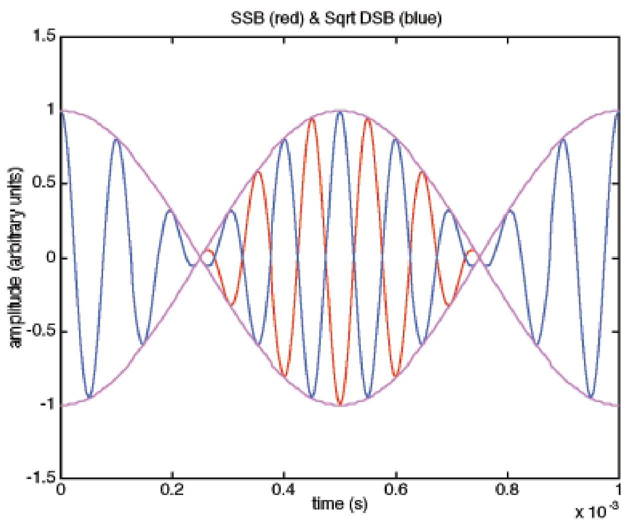


When this signal is squared by the process in the medium of air, you get back your original audio frequency of 2kHz, with zero distortion because no other frequencies are present. In other words, for a single-tone case, SSB gives a distortion-free signal with no pre-processing or additional signal conditioning.

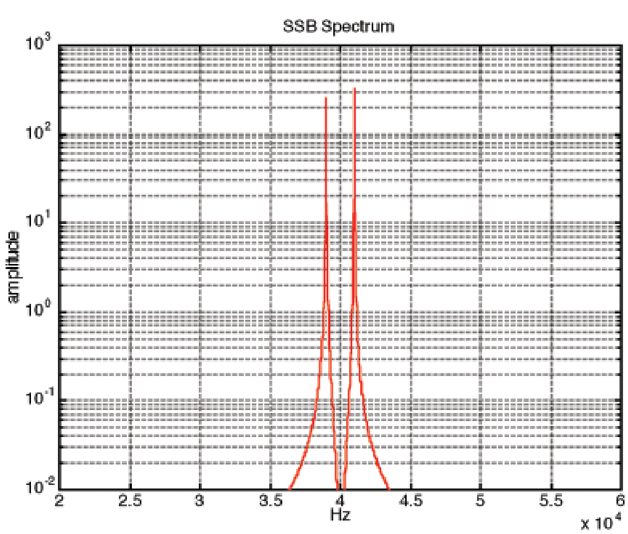
So in the case of no preprocessing, SSB is vastly superior to DSB. It has half the bandwidth requirement and no distortion compared to 100% THD.

As disclosed by the Japanese papers from the 1980s, the DSB signal can be pre-processed to eliminate distortion by running the audio signal through a square-root processor before it is utilized to modulate the carrier. This makes the envelope into a square-rooted sinewave. This comes with a price, though: bandwidth, in fact, theoretically, infinite bandwidth in both the upper and lower sidebands.

Lets look at a square-rooted DSB signal and an uncorrected SSB signal in the time domain:

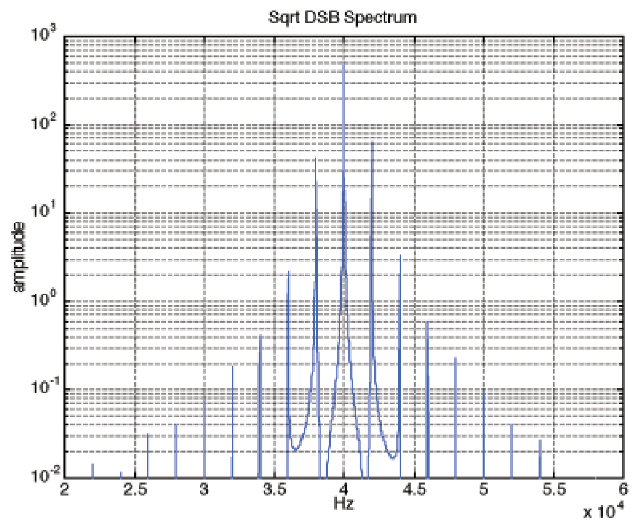


The blue DSB line lies on top of the red line when their values are the same, so when only the blue line can be seen, the red trace is underneath it. Both of these methods give the same envelope that will demodulate into a single, distortion-free audio signal (in this case, 2kHz). However, the spectrums required to generate these two envelopes are vastly different. Lets look at them. Here is the SSB spectrum with two frequencies:



For a 2kHz audio signal, only 2kHz of ultrasonic bandwidth is required.

Here is the DSB spectrum:

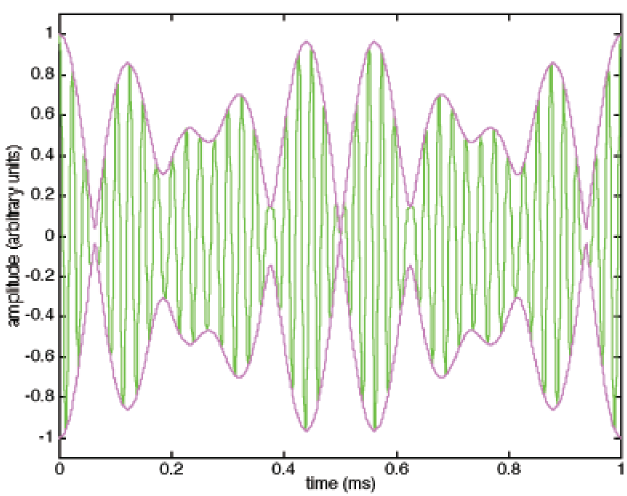


More than 30kHz of bandwidth is required to get the same ideal envelope to reproduce a mere 2kHz signal. For a 20kHz audio tone, the square-rooted DSB method requires some serious bandwidth that can't be realized in practice. In fact with a 20kHz signal the ideal DSB method would not only require an upper sideband of over 150kHz of bandwidth above the carrier, which is very difficult to realize, the least of which is building a wideband transducer. What is particularly problematic about this approach is the lower sideband would require distortion correction terms down into the audible range which would then become very audible as non-harmonically related distortion. The distortion correction terms themselves become distortion! A clear catch-22.

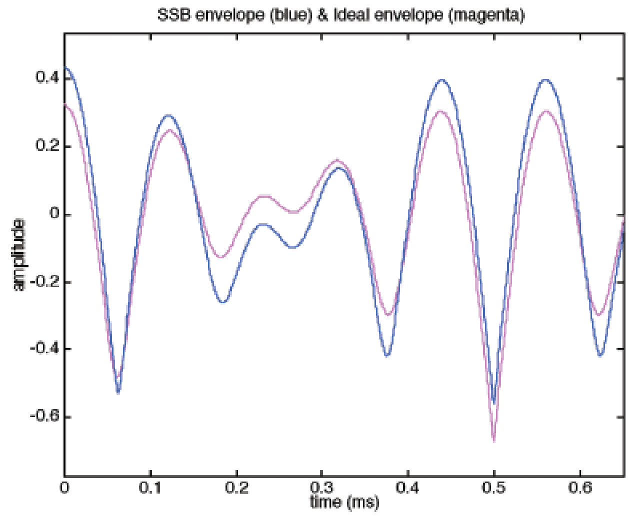
Fortunately, an effective square-rooted DSB system can still be made reasonably effective by truncating the bandwidth of the correction terms, even to the extent of not allowing any of the correction terms to fall outside the audio bandwidth. Theoretically, this would incur substantial distortion penalties and with equal amplitude signals it does. But, due to the nature of real world signals, the peak amplitudes of the audio spectrum will tend to fall in level for frequencies above 2 to 4kHz. Also, due to the inherent 12dB/octave high pass characteristic of the parametric output, substantial equalization will normally be used, at least down to some low frequency limit, and this will further reduce the high frequency output compared to the lower frequencies. These two things contribute to a very low modulation index for all high frequencies and therefore reduced distortion for those same high frequencies.

Even so, the distortion is still greater than what and ideal bandwidth could achieve and the single sideband system will always have less than half of the basic bandwidth of the processed double sideband approach.

So, let's look at some examples of the error terms that do occur when complex signals are used with a SSB system and how they can be suppressed. First, is a 40kHz carrier modulated to make 7kHz and 9kHz audio tones. SSB provides two upper sidebands with frequencies of 47kHz and 49kHz. The time domain signal looks like this:

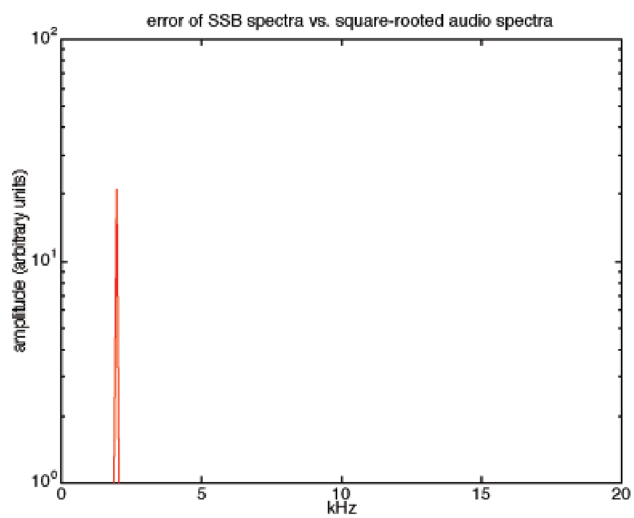


Next, the ideal envelope (square-rooted audio) for equal-amplitude 7kHz and 9kHz signals is shown on the same plot as the uncorrected SSB envelope:



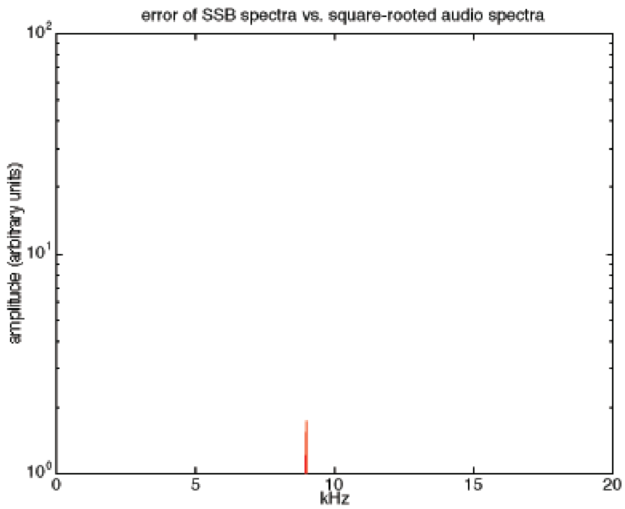
It is apparent that these two functions are similar, but not the same.

The resulting error spectrum (between the two envelopes) is shown below:



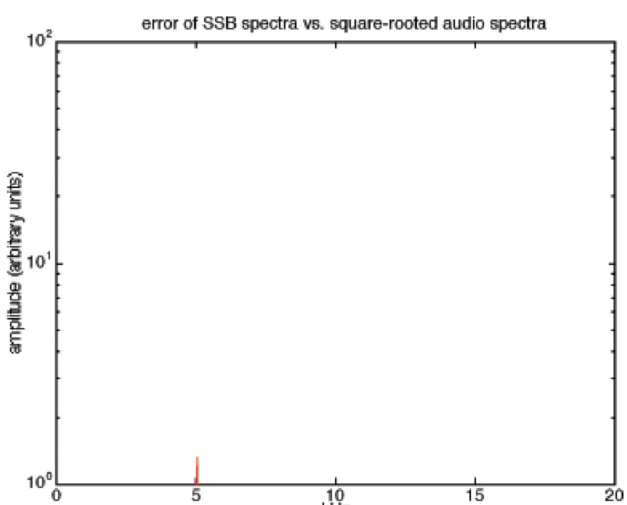
You can see that the only error term is at 2kHz. It can be proven mathematically that the only error term that results in the SSB envelope is in fact the 2kHz signal shown. However, if a 2kHz "correction term" (with the proper phase and amplitude) is injected into the modulator, the unwanted 2kHz error term can be suppressed completely.

Thus, a third sideband is added at 42kHz (in addition to the original 47kHz and 49kHz sidebands that normally occur) with the following results:



You can see that the 2kHz error term is gone, but a new one at 9kHz has appeared. However, the new error term is approximately 10dB lower in amplitude than the original 2kHz error term was.

If a 9kHz correction term (ultimately resulting in a modified 49kHz sideband) is inserted, with the proper phase and amplitude, into the modulator (in addition to the 2kHz, 7kHz, and 9kHz terms), the 9kHz error term can be suppressed completely, but a new term at 5kHz will appear at an even lower SPL:



This suggests that by utilizing a recursive scheme we can correct a SSB envelope without requiring any additional bandwidth. Note, as opposed to DSB, all of the correction sidebands fall within the bandwidth of the original audio input signal. Also, as can be seen, the amount of error occurrence, as a percent, falls with each recursive correction step used.

It is worth mentioning that a third modulation type has also been explored. We call this approach truncated double sideband. In this scheme a DSB multiplier is used and the lower sidebands are truncated with a filter and/or the transducer. The gives a hybrid between DSB and SSB. It allows for a simple multiplier but substantially retains the limited bandwidth of SSB. Moreover, it gives a controlled measure of self equalization to the demodulated audio.

For example, a DSB system with carrier at 30kHz could be high pass filtered at 28kHz so that it operated with a full 20kHz upper sideband but a truncated, 2kHz lower sideband. A 500Hz audio tone would result in upper and lower sidebands both being created equally around the carrier (for a high modulation index). A 5kHz tone would result in a strong upper sideband and a very weak lower sideband so the modulation index would be lower (the spectrum looks very similar to SSB). This lower modulation index results in a lower audio level for the upper frequencies. This serves to inherently equalize the audio for flatter response at lower frequencies.

The truncated DSB system can be used effectively with a recursive error correction scheme to eliminate distortion in the same way that the SSB scheme does.

A Further Novel and Advantageous Approach to SSB

Obviously, with SSB, one can choose to use either the upper sideband or the lower sideband. Generally, upper sidebands are used in SSB modulators.

In parametric loudspeaker use, the lower sideband (LSB) system has a number of advantages over upper sideband SSB or DSB, systems.

Ideally, double sideband systems require symmetrical outputs above and below the carrier frequency. Unless the transducer response is absolutely flat over at least a 40kHz range, preprocessed distortion correction will be less than ideal. Flat transducer systems are usually too low in efficiency to generate enough carrier output for parametric loudspeakers. To meet the requirement of frequency linear symmetry, equalized systems must utilize corrective factors that are linear with frequency rather than logarithmic, which is very difficult to realize, so even a smoothly peaked transducer will not be linearly symmetrical above and below the resonance/carrier frequency.

Therefore, the theoretically ideal square rooted double sideband system cannot be fully realized.

Transducer response is very predictable when transducers are used below resonance. In this range they operate in their stiffness mode and for a critically damped system the high pass characteristic is consistently 12dB per octave. For underdamped systems, with much greater efficiency at the resonant frequency, the high pass characteristic is greater than 12dB per octave down to a given frequency after which it shifts back to 12dB per octave down to dc which, as will be seen later in this paper, more closely matches the ideal equalization for ultrasonic to audio parametric conversion. For the distortion correction systems to be effective they depend on the linear response of the transducer to be a simple function to achieve significant distortion reduction. The stiffness controlled, high pass function of most transducer types is quite predictable and repeatable.

Upper sideband, above resonance, mass controlled region frequency response of most transducers can be somewhat erratic.

Further, the lower sideband has program material peak energy factors that fall with frequency. This means that in a 20 kHz bandwidth, the sideband information that is displaced 20kHz from the carrier will be very low level. If we were to use a 40Khz carrier for LSB, the spectral content of the program material would guarantee significantly reduced output at all frequencies below 30kHz, with the spectrum falling between 3 and 6dB per (audio) octave below 38kHz depending on program type. Further, the parametric conversion process demands another 12dB per octave of roll off with ascending audio or descending LSB ultrasonic. This results in at least a 15dB per octave "audio" attenuation which translates to more than 90dB per octave from 40kHz down to 20kHz in the ultrasonic. (This assumes a 300Hz to 20kHz audio bandwidth.)

Below resonance/carrier in a LSB system the saturation levels and attenuation levels are minimal with decreasing ultrasonic and ascending audio frequencies. In an upper sideband system, be it SSB or DSB, above resonance/carrier the ultrasonic attenuation and saturation levels both increase with frequency.

Even though the DSB systems have outputs that can descend down into the upper audio frequencies, that output is greatly attenuated since it is the mirror of the high frequency audio. Even though the upper SSB system has no lower sideband, which should allow a lower carrier frequency, the carrier is at very high levels so it may still need to be placed at a higher frequency than just above the audible range.

Band limited, in band corrected, LSB systems should provide the best of both worlds with the potential for greater output and much more effective distortion reduction.

Also, there are many higher efficiency transducer techniques, such as tuned pipes, that increase output at the fundamental resonant frequency but cause many problems with frequencies above that range. Interestingly, these high efficiency, resonant devices remain well behaved below resonance. This implies, that if the upper sideband is eliminated, there may be further improvements in efficiency available in emitter design without sacrificing linearity.

Lower sidebands are also be better because the low audio frequencies are at higher ultrasonic frequencies and therefore have greater directivity associated with them and visa versa for high audio frequencies, further helping to maintain high directivity at low audio frequencies.

Because attenuation and saturation effects increase with higher frequencies, minimizing any significant upper sideband extension requirement is significant in achieving higher performance in a parametric array.

Ultimately, the narrower the bandwidth the greater the system efficiency can be.

Besides being effective from a distortion reduction standpoint, as will be seen in the next section, ATC's proprietary, narrow bandwidth, recursive, lower single sideband system can provide much greater parametric output by interacting more effectively with the associated ultrasonic transducer.

Utilizing the information disclosed above, it can be seen that the system that provides significant advancement over the prior art, while eliminating preprocessing side effects, is a single sideband processor utilizing a square rooted envelope reference to calibrate a recursive, zero bandwidth distortion canceller operating as a lower sideband modulator. This is the basis for the proprietary parametric processor currently being implemented at American Technology Corporation.

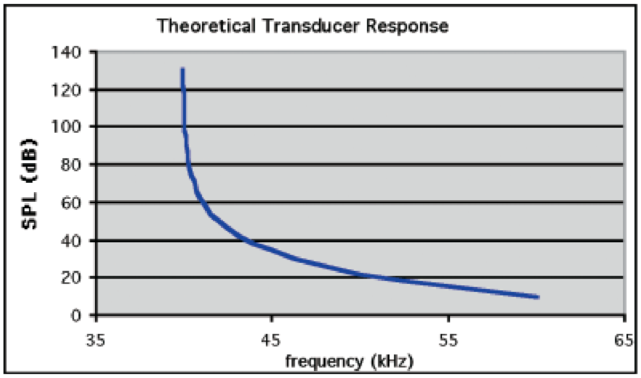
Transducer Technology

General Requirements

In order to make a parametric loudspeaker work, ultrasonic energy must be emitted into the air. Electrical signals are converted into these acoustic signals by means of an ultrasonic transducer.

Acoustic transducers or emitters can designed to cover a certain frequency range, and to have a certain dispersion patterns. The optimum parametric emitter would have bandwidth from around 20kHz to infinity, and a sharp dispersion pattern (one that gives a collimated beam of ultrasound), and unlimited output capability. Unfortunately, this is not physically possible to achieve. What we have to shoot for is 20kHz of useable bandwidth (for use with SSB modulation giving 20kHz of audio bandwidth), a resonant peak where the carrier will be placed, and a falling output level with frequency to provide a measure of self-equalization in the system (to overcome the 12dB/octave roll-off of the demodulation mechanism).

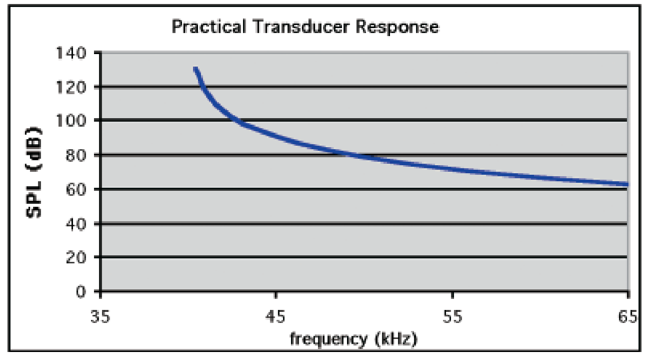
The optimal slope that the frequency response should fall off can be computed by applying Berktay's solution to see what ultrasonic signals are needed to give uniform audio response. In order to achieve a flat audio response from 20Hz to 20kHz, the upper sideband level must fall by 20*Log(4)dB or 12.04dB per audio octave. Here audio octave means going from an upper sideband who's frequency is, say, f0+1kHz to an upper sideband who's frequency is f0+2kHz. For example, if the carrier, f0 is 40kHz and 120dB and the upper sideband is 41kHz and 120dB too, a 1kHz audio signal will appear at a given level. To have the same level appear for a 2kHz tone, the carrier can be left alone, and the upper sideband needs to be changed to 42kHz and 107.96dB. So in order to equalize the levels of the entire audible range (20Hz to 20kHz), the following transducer frequency response would be required:



Notice that the level difference between the carrier at 40kHz and the 60kHz upper end is 120dB. Since the audio range is 10 octaves wide, severe equalization is needed to achieve uniform response. Equalization generally means bringing peak output levels down to match the lowest levels present. The toughest sound to make is the 20Hz tone, so the transducer would be driven at its maximum level when making this tone. All the other tones are generated more efficiently, so less modulation level (meaning less upper sideband power) is used to bring these other tones down in level to the achievable level of the 20Hz.

Such an arrangement would give very poor audio output levels. If the transducer could be driven harder, say to 200dB at its resonance, one would think the performance should improve dramatically. It would not. The air column in front of the emitter would be "saturated", and much of the energy would be dissipated as heat, rather than turning it into sound. With the current state of the art in parametric loudspeakers it can be seen that the lowest audio frequencies are too inefficient to generate in a practical manner. If we need to have them, it is currently, best left to a conventional bass module.

Now, if we give up on trying to generate equal-level signals below, for example, 500Hz, we have a much better chance of providing good output over the rest of the range. The frequency response of a transducer designed for 500Hz to 20kHz flat audio response would look like the following:

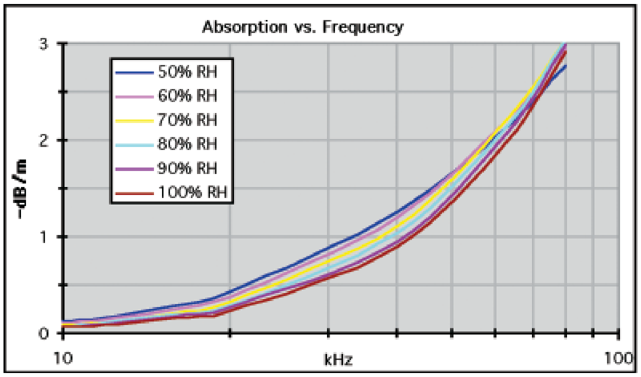


As you can see, this is much more realistic. Because the upper sideband levels are so much higher, the overall performance will be much better. In fact, the output levels will be increased by more than 56dB over the band of concern. Such an emitter is realizable. Also remember that there will be output below 500Hz, just not at the same level as the rest of the bandwidth.

As mentioned earlier, a collimated beam is a must. Conversely, with a point source the inverse square law holds. This means that the intensity and SPL drop by ≈ 6dB for every doubling of distance from the source. This is because the wavefronts are expanding spherically around the source, so the intensity falls as the surface area of the sphere grows (intensity is power/unit area). With a plane wave source (where the radiating surface diameter >> a wavelength being emitted), the wavefronts do not spread appreciably and a collimated beam results. The only losses in intensity occur due to molecular friction. The attenuation is gradual over distance. For example, at 68° F and 80% relative humidity, a 40.0kHz signal loses 3.1dB in ten feet. This attenuation grows with increasing frequency so lower operating frequencies are desirable for minimizing losses.

Values Given in -dB/m

Freq (kHz)	50% RH	60% RH	70% RH	80% RH	90% RH	100% RH
6.3	0.05131	0.04251	0.03673	0.03272	0.02984	0.02772
10	0.12307	0.10184	0.08733	0.07697	0.06932	0.06349
12.5	0.18699	0.15567	0.13369	0.11774	0.10581	0.09663
16	0.29387	0.24778	0.21401	0.18889	0.16978	0.15490
20	0.43532	0.37378	0.32592	0.28911	0.26051	0.23793
40	1.25533	1.19537	1.11242	1.02836	0.95158	0.88428
63	2.14711	2.22064	2.20963	2.14933	2.06542	1.97362
80	2.76840	2.95171	3.03230	3.03586	2.98902	2.91301

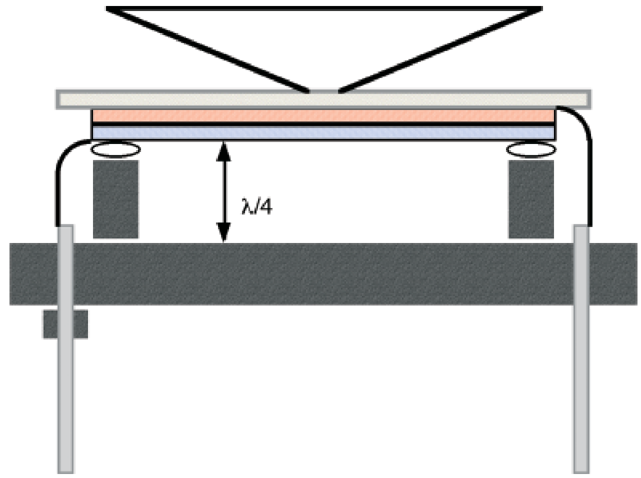


(For the full table of attenuation values vs. frequency, see section 14 in a CRC Handbook of Chemistry & Physics.)

It also necessary to make the emitter about 0.5ft^2 in surface area to ensure that there is a sufficiently large ultrasonic column. Berktay's solution says that the demodulated level will be proportional to the area of the beam. Using a broad beam is also a way to avoid saturation. Efficiency is gained by spreading the output power over a larger area. Since a large area and the short wavelengths needed are not generally compatible requirements, an array of small elements is generally used to get both.

Other emitter essentials include reliability over a 5-10 year life span and manufacturability, reasonable cost, and a high output capability (>140dB re. 20μ Pa at resonance).

The design used in virtually all of the prior art before 1998 is based upon a PZT "bimorph". This is a two-layer wafer of poled PZT material arranged in such a way that it bends from concave to convex as the AC is applied. This gives large displacements in the center of the element. To couple to the air load and give the PZT a mechanical load, a lightweight cone about 8mm in diameter is attached to the center of a load matching plate which is bonded to the bimorph. The cone has a fundamental vibration mode at the bimorph's loaded resonance. To produce more output, the bimorph is mounted over a l/4 cavity. See the drawing below.



When a voltage is applied across the pins (that are protruding downward), the red element gets longer while the blue one shortens, causing a bend in the bimorph. When the polarity changes, the opposite bend occurs. The maximum displacement change is in the center of the element where the cone is attached. This way, the cone undergoes maximum displacements. Also, the outer edge of the cone moves as much as three times further than the inner portion as it flexes.

These devices are good for many applications, but their parametric ability is limited. The frequency response is okay for approximately 10 kHz above resonance (a sloping roll-off above 40kHz), and the efficiency is not bad. The total output is good also (123dB for the Nicera AT40-12P at 30cm with 10Vrms drive). Unfortunately, they are very small (about 7mm in cone diameter) so nearly a hundred have to be used to make a decent array with enough beamwidth for effective parametric conversion. Worse, they are fragile and when driven to the required levels, a few in a given array begin to squeal and create subharmonics almost immediately. This seems to occur because the cone flex required for high-output results in fatiguing and cracking around the center weld spot.

Also, there is a large electrical phase shift a resonance. This would not be problematic, but each device has a slightly different peak frequency which limits their arrayability. If a carrier is set at the average peak frequency of a group of emitters, many will be out of phase with their neighbors. Because of this, it was found that they work best when the carrier frequency is set slightly off-peak value.

Advancements in Parametric Emitter Development

It was clear that there was no “on the shelf” transducer available that would allow this technology to reach its potential. In light of the dearth of prior art transducers for effective use in parametric loudspeakers early on ATC began research into new types of transducers that would advance the performance of parametric systems.

Early experiments produced very high output devices, capable of producing over 155 dB of output at the ultrasonic carrier frequency. These transducers were fairly small, with a diaphragm diameter of approximately 30 mm. It was quickly discovered that even though there was significantly more output than any previous device the parametric conversion was very weak.

In referring back to the Berktay solution it is found that the output is proportional to the column area. While this would seem obvious, as any loudspeaker output is directly related to radiating surface area, greater surface area plays a much more significant role in parametric systems. Because of the inherent parametric output restrictions, due to the saturation of the medium at high intensities, a larger ultrasonic emitter with less output per unit of radiating area will significantly out perform a smaller device with greater output per unit area.

Various experiments were performed with variations in packing density of individual PZT bimorphs and it was found that much greater conversion efficiencies were available from lower density configurations such as an open ring topology, which was found to have significantly greater parametric output per driving area while using less than half the devices to create a larger diameter ultrasonic column.

What was needed was a transducer that did not have the inherent limitations of the PZT bimorph and would further lend itself to large area parametric column generation. With this in mind we moved away from individual, high output devices and explored large, monolithic, thin film transducers topologies based on electrostatic, planar magnetic and piezoelectric films. At ATC there is ongoing research into many types of ultrasonic devices, but of the thin film transducers, the piezo film generates the greatest ultrasonic output per unit area while providing easily scalable singular structures of any diameter desired for a given application.

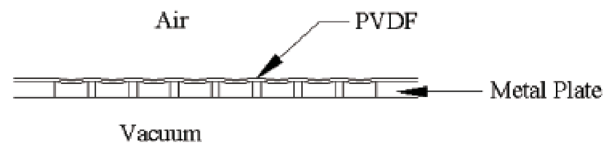
Piezoelectric Film Parametric Transducer

The most active piezo film is polyvinylidene diflouride or PVDF for short (also sometimes referred to PVF2). This film is commonly used in many industrial and chemical applications and for such applications, the raw film is used. In order to be useful for ultrasonic transduction, the film must be polarized, or activated. This is done by one of two methods. One method yields a “uniaxial” film that changes length along one axis when an electric field is applied through it. The other method yields a “biaxial” film that shrinks/expands along two axes. Finally, the film needs to have a conductive electrode material applied to both sides in order to achieve a uniform electric field through it (by having the same potential at all points on one side).

The first proof of concept devices created for parametric use utilizing this material were completed in late 1996 and the initial results were very promising.

Piezoelectric films operate as transducers through the expansion and contraction of the X and/or Y axes of the film surface. For use as an emitter rather than a sensor or receiver, the film will not create effective motion in the Z axis unless it is curved or distended in some what so that the expansion and contractions can be converted into Z axis movement and create displacement generating acoustic output.

In one of the simplest implementations of this concept, one takes a sheet of PVDF and lays it over a metal plate with an array of holes in it. One can then apply pressure or vacuum to one side of the plate to create an array of PVDF diaphragms, each with the diameter of the hole under it, which are under uniform tension and can be driven in parallel. A schematic cross section of such a device is shown below.

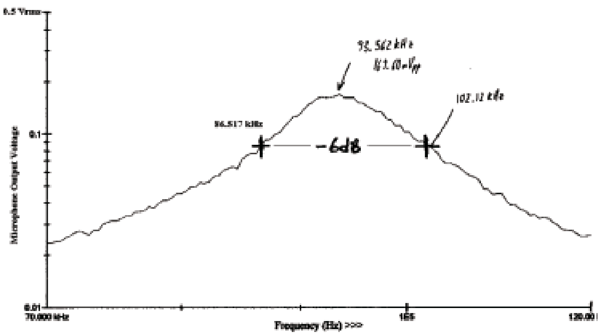


To achieve a resonant frequency corresponding to a parametric carrier frequency, distended piezo film emission areas were created by placing the film over a plate containing an array of openings that were sized for a 40 KHz resonant frequency and a differential pressure was applied to the surface facing into the plate of openings. It was calculated that for a 40 kHz resonant frequency, with a 14.7 psi pressure differential, a hole size of 0.140" was needed. A vacuum was used to distend the film instead of a positive pressure so as to eliminate back wave effects.

The first device to be built was 1.75" in diameter with 85 9/64" holes (selected to use a standard size drill bit) arranged in a tight hexagonal pattern with center-to-center spacing of 0.160". 28 micron film was used and with nearly a full vacuum behind the film, the resonance frequency was 37.23 kHz. The output was 136.5 dB with 73.6VPP drive. A very encouraging result for a first try.

Because of the flexibility in calibrating the operating resonant frequency, we could explore a wide range of frequencies for parametric operation. A number of experimental devices were built with operating center frequencies ranging from 25 kHz to over 100 kHz. (Using specialized emitters parametric arrays have been tested at ATC up to frequencies greater than 500 kHz.)

The measured results of a typical, early proof of concept device showing a resonant frequency of approximately 93 kHz is shown below. This device utilized 25 micron PVDF film on a plate with 1 mm holes with a back pressure of 1 atm.



In ATC's early days of developing ultrasonic emitters with PVDF there were a number of issues to over come, such as, containing a vacuum, unit to unit variation control, selection of metalization material and optimum processing of the piezo film for maximum output and reliability. Through the use of a new type of proprietary PVDF film production process, the current emitter is a stable, repeatable and very practical device to manufacture.

Through the use of differential pressure distended PVDF film, we now have the first purpose built emitter for use as a parametric loudspeaker. With this device a solution now exists to solve the prior art parametric transducer problems with an ultrasonic emitter that:

- eliminates multi-transducer unit -to-unit variability through the use of a single monolithic, thin film for the entire array
- exhibits very high efficiency at the carrier frequency with attenuated, self equalizing slopes at the sideband frequencies
- has an adjustable resonant frequency adaptable to various carrier frequencies and modulator types
- eliminates out of band (audio range) sub harmonics of the prior art devices
- has ultrasonic bandwidth at least equal to that needed to reproduce the widest band audio
- a monolithic structure for matched output over the active surface with repeatable, simplified construction
- has greater than 140 dB ultrasonic output capability
- has inherently low distortion

When matched with the zero bandwidth parametric processor, a practical, reliable, advanced parametric loudspeaker system of low distortion can be realized for a wide variety of applications.

Parametric Loudspeaker Facts and Limits

- Maximum parametric output throughout the audio range is always slightly less than the maximum volume velocity of the emitter at the carrier frequency. i.e. 12 dB reduction per descending octave in output relative to primary, ultrasonic frequency output
- Parametric output is proportional to the area of the ultrasonic column
- When used below saturation, the parametric output increase is proportional to the square of the increase in carrier level
- Ultrasonic directivity is based directly on emitter diameter
- Parametric directivity is based indirectly on the ultrasonic directivity and directly on the length of the ultrasonic column or virtual end-fired array
- Lower modulation index reduces distortion
- Greater modulation index increases parametric gain
- Single Sideband envelope is equal to square rooted envelope for a single tone
- Emitter linear errors (ultrasonic amplitude response) can translate to parametric non-linear errors in any error correction scheme
- Square root correction is needed less for high frequencies in real world program material due to:

  • Spectral power attenuation above 2kHz causing lower modulation index and therefore lower inherent distortion

  • Harmonic distortion for signals above 7 to 10kHz tend to be inaudible
- Saturation sets in at levels 6 dB lower for each octave of increased ultrasonic operation
- Output volume velocity is decreased 12 dB for each octave of increased ultrasonic operation
- Human Factors:

  • Studies of intense airborne ultrasound exposure so far have found that any noticeable effects of ultrasonic outputs have been due to the unwanted high-frequency audible signals that can accompany ultrasonic equipment and not the ultrasonic signals themselves. The unwanted audio frequencies can be essentially eliminated with zero bandwidth correction schemes combined with using emitters that do not generate sub-harmonics.

  • For any heating effects to occur, the parametric systems would have to operate at more than ten times the power output of an effective parametric loudspeaker.
- For ultrasonic levels above 135 dB, at 40 kHz, saturation of the air creates diminishing returns relative to the effectiveness of any further increase in ultrasonic level achieving significant increases in parametric output

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