New Test Signal, Phi Low-High Split Band Spectral
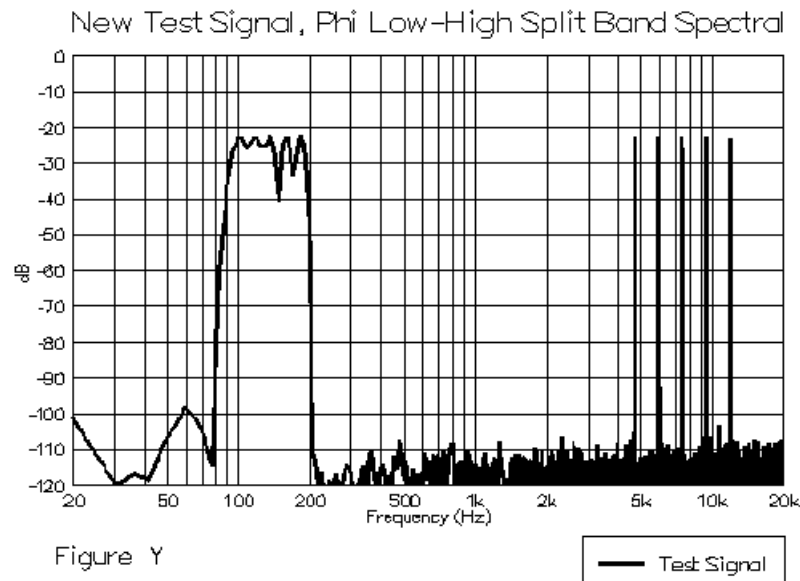
Figure Y

Test Signal

The New Test Signal

Phi Spectral Contamination

My new test signal is as much a concept as it is a specific set of frequencies put together in a certain way. Traditional test signals either use a single sine wave tone or at most, a pair of tones, to explore simple harmonic distortion, and simple IM distortion. Music is composed of many tones, all happening at once, and with varying relationships. - - Multitone test signals use more than one single sine wave, more than just a pair of sine waves, in order to more thoroughly excercise the system. Before I came up with the new signal, the existing multitone signals as used, and as available via major hardware and software had some common faults. One of these was the use of regularly spaced tones, with spacing intervals that tended to be the same.

For instance, one of the popular test signals had tones at 100, 200, 300, 400, 500, 700, etc and at other frequencies as well. The problem with this is that the spacing for a lot of the frequencies is the same: 100 hz. This means that IM distortion generated from these particular frequencies would tend to have as one of the common frequencies a product at 100 Hz (200 -100, 300 - 200, 400- 300, etc.). Not only would all of these tones tend to have a similar distortion product frequency, but it would be impossible to even determine the composite level of these particular distortions, as the 100 Hz tone present in the test signal would overwhelm them. In effect, the measurement would be blind to these particular distortions.

I think that it is easy to see that there are many possible combinations that will end up covered up by the primary test signal tones, such as distortion products at 200 Hz, etc. There is also the fact that many harmonics will be covered up too. The harmonics of the 100 Hz tone at 200, 300, 400 and 500 Hz will all be impossible to detect or measure.
One of the current multitone test signals has tones spaced every one/third octave, which guarantees a lot of cover up of IM and harmonic distortions with this particular test signal. Not only that, it has 31 tones, which means that the overall level of each of the tones has to be kept low enough ot avoid clipping the system, whether it is the analyzer, the device under test, or whatever would be stressed. For 31 signals, this is about 30 dB down from 0 dB nominal.
After checking out what was out there, I realized that if you make all the tones fall at intervals where they would not have the same spacing ratio's, and would not have the same frequency

intervals, this would avoid the cover-up of the distortion products. This idea in and of itself turned out to not be new, but after checking around, I determined that no one had come up with a consistent method to try and assure that as few of the tones, and all of their distortion products up to higher orders, would not interfere with one another, and not be covered up by the original tones. If you select frequencies at random, there could always be a bad combination, and it would not be easy to predict.

What I wanted was a spacing ratio that would make as many of the products spaced away from one another and from the original tones as much as possible. Eventually, I found that the ratio known as Phi, or the Golden Ratio, did the trick. I started out with simple multitones, and gradually developed several different multitones based on various permutations of the number Phi, including submultiples of useful ratios added to integer numbers.

Besides the use of same spacing intervals, and integer spacing ratios, one of the original multitone test signals, the one that coined the term spectral contamination, used either entirely or primarily out of the audio band ultrasonic signals. The idea here was to stress the component under test and see how it reacted to this stress in terms of in band audio distortions. It also used same frequency spacing with the attendant problems. -

Out of all this came a core of useful test signals that all have their place in terms of sensitivity to various effects and distortions.

There are two basic types: evenly spaced tones using the same Phi multiplier on each subsequent tone, and split band types which concentrate a set of tones at various portions of the spectrum, leaving large areas of the audio spectrum free from stimulus tones. In both cases, the IM distortion products, the cross modulation products, and most all of the harmonic products are free from cover up by the original tones, and free from stepping on one another.

For the distortion products at higher orders that do come within a closer spacing that can easily be distinguished on a spectrum analyzer, there is a simple technique that allows positive determination of the source of the distortion. Again, this is something that was not only impossible with some of the other multitone methods, but was made practical for the first time due to the avoidance of multiple and close stack ups.

Final Test Signal Versions

In addition to working out this spacing relationship, I also came up with a spreadsheet that allows easy determination of the source of a particular distortion, which when used in conjunction with the technique mentioned above, allows you to exactly determine which two (or more) frequencies were the source for that particular distortion product. Final Test Signal Versions Here are the the final versions of the test signals I developed:

Phi 6 Spectral: 100, 261.8, 685.4, 1794.4, 4697.9, 12299 Hz. Individual level of each tone is at -15.6 dB.

Phi 12 Revised Spectral: 100, 122.0, 261.8, 348.2, 685.4, 987.0, 1794.0, 2870.4, 4697.9, 6765.0, 12299, and 16358 Hz. Individual level of each tone is at -21.6 dB.

Phi Low-High Split Band Spectral: 100, 116.18, 134.98, 156.80, 182.19 and 4697.9, 5927.8, 7479.7, 9437.9, 11909 Hz. Individual level of each tone is at -20 dB.

Phi Low-Mid Split Band Spectral: 100, 116.18, 134.98, 156.80, 182.19 and 986.99, 1245.4, 1571.4, 1982.8, 2502.0 Hz. Individual level of each tone is at -20 dB.

Phi Tri-Band Spectral: 100, 116.18, 134.98, 156.80, and 986.99, 1245.4, 1571.4, 1982.8, and 6764.9, 7618.5, 8579.8, 9662.5 Hz. Individual level of each tone is at -21.6 dB.

For my tests, I generated the tones in the digital domain, burned them to a CD-R, and used a squeaky clean CD player as the test signal source. Analysis was performed on a 16 bit FFT based system, using 8K point FFT's. This results in 4096 real data points within the audio band, resulting in an approximately 5 Hz bandwidth for each FFT frequency bin. Other instrumentation for FFT spectrum analysis is available that is capable of more than 16 bits, with resolution up to 64k FFT's, which makes the FFT bin width about 0.6Hz wide. Some of these cost a fortune, and others as little as a current PC with a studio grade sound card and the appropriate software.

During the course of testing, it was found that CD players had measurable amounts of IM distortion, and all measured different using these test signals. So much for all CDP's are alike. I used a CDP that had near source signal levels of output distortion so as not to compromise the dynamic range of my measurements. To my knowledge, no one has published anything about these IM distortion differences, probably because they do not tend to show up with traditional two-tone IM tests as well. One player showed lower SMPTE IM and harmonic distortion levels, yet had significantly higher levels of Phi Spectral IM distortion than another player. Many other measurements were made as well, a very high quality cassette deck was compared to a mini-disc deck; digital electronics were tested and compared, raw loudspeakers and loudspeaker systems were tested and compared. Each one of these audio components reacted in their own specific fashion to the test signal, and in most cases, levels of the traditional IM and harmonic distortions were inconclusively low between the various component test pairings. Sometimes one unit would look significantly better using traditional measurements, yet end up measuring much worse using the multitone. The correlation between which unit sounded good, and which didn't followed the Phi Spectral measurements almost exactly.

Of course, there was one other audio component that was measured: speaker cables.

Check out other Spectrals

A quick note: one of the current spectral contamination measurements is currently being used at Mix magazine for studio monitor reviews.

Check it out at : http://www.mixmag.com/ and click on the "Tannoy Reveal 2-Way Reference Monitors" link, and go down to Fig. 4. Note the number of tones at even frequency intervals.

One other thing about my version of Phi Spectral signals. They have a very high crest factor, on the same order as the amount of individual tone reduction factor, or a little less. That is about 15 to 21 dB of crest factor. This is also more like real music, instead of a single sine wave with it's 3 dB crest factor.

Testing Speaker Cables with the PHI Spectral

Now that I have covered a little bit about the test signal itself, I will go into the actual cable tests. Explanation of Measurements In measuring cables, I hypothesized that for bi-wired speaker cables, the big difference would be in the current flow, as the impedance's of the crossover would create a preferential situation with regard to the flow of current in the woofer cable vs. the tweeter cable. To check this, I used a Pearson Electronics Model 411 current sensor (http://www.pearsonelectronics.com/) to measure and analyze the current flow in the speaker cables. No other method of measurement will yield the same information, as a series resistor to convert the current to a voltage will introduce it's own resistance, tending to swamp any speaker cable differences out. Voltage measurements will not provide the same information, unless you are using a resistor load, without crossovers and actual speakers involved. I initially tested a single 12 gauge zip cord against my cross-connected 89259 in bi-wire mode, that is what was published in my AES preprint. The deadline for that paper was June 15th, since then, I have tested various combinations of gauges, various retail audiophile cables, all of which have been tested in bi-wire and single cable modes.

While I have found differences between single speaker cables of 3 and 4 dB between the various cables in single wire mode, which I believe to be repeatable and consistent, but these small changes would no doubt be questioned endlessly by certain parties, so I make no definite claims for these results, other than to relate them for future study. Perhaps a 20 or 24 bit analyzer with the appropriate resolution can quantify these cable differences more clearly. Gauge did not strongly influence the distortion levels, the difference due to smaller and larger gauges were well within the measurement error limits, while the differences between audiophile cables and zip cord were generally higher, although not above a level beyond any doubt, as I have indicated above.

Analysis

When I obtained the distortion plots for the bi-wire vs. single wire conditions, I was careful to analyze the results, so as to fully understand what was going on. At first, I felt that the reductions might be due to the simple aspect of the crossover impedance's, and that perhaps the distortion reductions followed the crossover curves. Close examination revealed that not only does the distortion plot NOT exactly follow the curve of the crossover current flow (this can be examined by looking at the web site, page7 vs. page 8, and comparing specific frequencies. I posted a set of example frequencies and discrepancies earlier in response to concerns about the crossover roll off being the only valid effect here.) but for the Phi Spectral signal, the crossover roll off DOES NOT DIRECTLY AFFECT THE DISTORTION PRODUCTS AT THAT PARTICULAR FREQUENCY because they are not generated by any frequencies at that location in the frequency domain, but by the bundle of tones at the low end and at the high end! The distortion products would not be affected by the current flow (or voltage) frequency response at that point on the graph!!!! In order to make this perfectly clear, let me give a concrete example. For instance, the distortion spike at just above 1 kHz in Fig. Z on page8 at the web site. It is about 25 to 27 dB higher for the single cable than it is for the bi-wired tweeter cable. Now even if we completely ignore the fact that this distortion product derives from a pair of original tones that are either in the 100-200 Hz band, or in the 4.7 to 11 kHz band, and pretend that it is tied precisely to that point in the frequency domain, there is something inconsistent with the accusations made: the current flow for just the tweeter cable is only down 15 dB from the peak current flow, not 25 dB If we look at the current flow for the single wire cable, it is at a level that is only 4 dB higher, not 25 dB. I will post this graph, to be called Fig. W, on my web site at page9.

New Test Signal - Measurement

I will repeat this: the entire band of distortion products from 200 Hz to 4.7 kHz in Fig. Z DO NOT LIVE AT THOSE FREQUENCIES, they exist because of IM between two other frequencies far removed from that locale, and any local deviation in FR is not tied directly into their levels! Remember, we are looking at the current distortion IN THE CABLES, not anything else at this point.
I will now go over exactly how the distortion manifests.
How can the distortion in the speaker cables manifest to our ears? The woofer only has significant acoustic output up to about 9 or 10 kHz, while the tweeter only has significant acoustic output down to about 1 kHz. Any distortion that would be audible could only come from output below the 9 or 10 kHz that the woofer can have output at, and above 1 kHz for the tweeter.
The crossover IS rolling off the drivers output, there is no doubt about this. What may get overlooked, is that any distortion reduction can only come between the acoustical roll off and the crossover roll offs for each driver. There is indeed a limited range where the distortion will manifest for each driver.

For the tweeter of this particular speaker system, under these conditions, that range is from 1 kHz to about 6 kHz, and for the woofer, that range is 6 kHz down to about 3 kHz, with the crossover at about 3 kHz for this system. At 1 kHz, the crossover voltage tweeter output will be down approximately 20 dB, and approximately 10 dB at 3 kHz. For the woofer, it is down 10 dB at 3 kHz, and 20 dB at 6 kHz.

The crosover roll off itself may be considered as part of the reduction, but this roll off is applied to both the signal coming from the tweeter bi-wire speaker cable AND to the single speaker cable, so that any advantage this represents will be present for both cables equally. So the amount of reduction in distortion is maintained, despite the roll off inherent in the crossover.

If all the various factors are taken into consideration, as I have explained them, the amount of distortion reduction could be said to be about 15 dB in a region from about 2 kHz to about 5 kHz for the tweeter, and for the woofer, a difference of about 3 to 5 dB in a region from 3 to 6 kHz or so. There may be lesser reductions outside these bands, and this is not taking into account other factors which I feel might make these levels, regions and amounts more significant than they seem at first glance.

So the actual amount of distortion reduction is not actually 20 dB, even though there were regions on Fig. Z that had 25 dB differences at frequencies that might have been relevant with a casual analysis. It is more like a range from about 15 dB to 3 dB over a frequency band from 2 kHz to 6 kHz in terms of what would possibly be audible.

Note that these levels and range is just for this particular test signal, some of the other Phi Spectral test signals gave different results, being based on different frequency mixes. The Phi Tri-Band for instance gave a wider frequency range, while the Phi 12 Revised gave a few isolated level differences that were at the 15 dB point or more, all things taken into consideration as above.

Tech Details; Drive level for Fig. Z was 2 VRMS into the 4 Ohm nominal load the speaker system presents, so that the RMS level of the signal was at 1W, due to the crest factor, the signal peaks would be hitting 80W or slightly more. 0 dB on the graph equals 1 ampere. The noise floor of this particular measurement is approx. -90 dB.

Repeatability of the measurements made using the current probe, the 16 bit FFT and the entire measurement chain were fairly consistent, with up to a maximum of 1 to 2 dB of variation in the higher frequencies, and up to 2-3 dB of level variations in the lower frequencies. Frequencies below the 100 Hz test band tended to vary more due to the FFT bin size and LF hum and noise.

Bi-Wiring Measurements

Current Flowing To Woofer & Tweeter Sections

Figure X
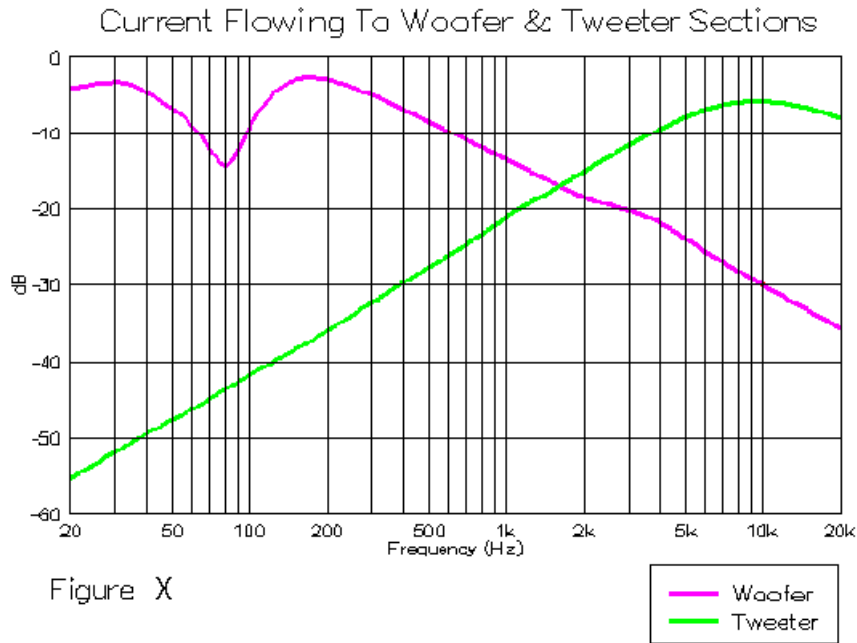
Woofer
Tweeter

Current Flow Through Bi-Wire Speaker Cables
Fig. X is a graph of the current flow through the two different cables in a bi-wire hookup. Note that the woofer cable carries mainly lows, and the tweeter cable mainly highs. - - - - This is what accounts for the advantage bi-wiring has over a single cable, the current flows for the highs and lows are separted, and the resultant magnetic fields are separated too.

Phi Spectral, Single vs. Bi-wire Tweeter Cable

Figure Z

Single Cable Current
Bi-Wire Tweeter Cable
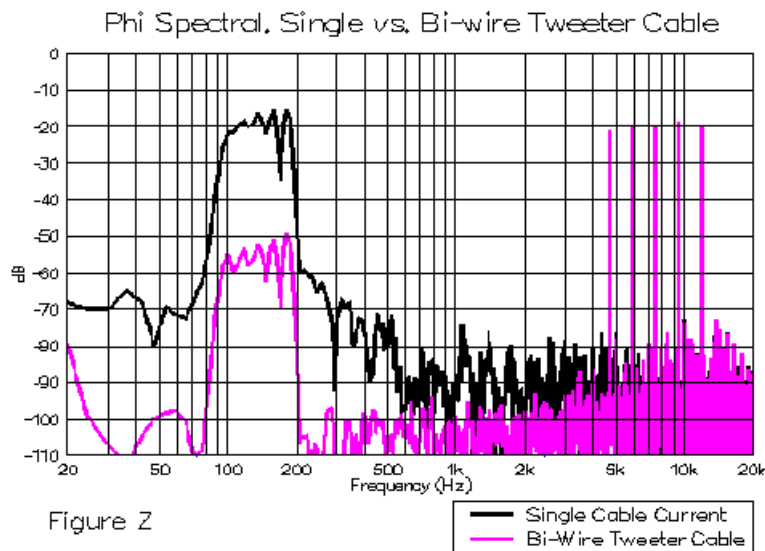
Single Speaker Wire vs. Bi-Wired
This graph shows the reduction in IM distortion products when using two separate speaker cables to the separated crossover sections of a speaker, otherwise known as bi-wiring, compared to using a single speaker cable. - - - - Only the single cable vs. the tweeter cable is shown here for clarity.

The graph depicts curent distortion products.
The distortion products depicted are detected using a current probe, and consists of current distortion in the wire. - - - The current is what moves a dynamic speaker, and is involved with driving most electrostats via a step-up transformer. - - - - Current produces a magnetic field, and my theory is that it is the reduction of the intermodulation of the low frequency magnetic field with the high frequency magnetic field that provides the great reduction in IM distortion products measured.

Current Flowing Into Entire Crossover

Figure W

0 dB = 1 Amp
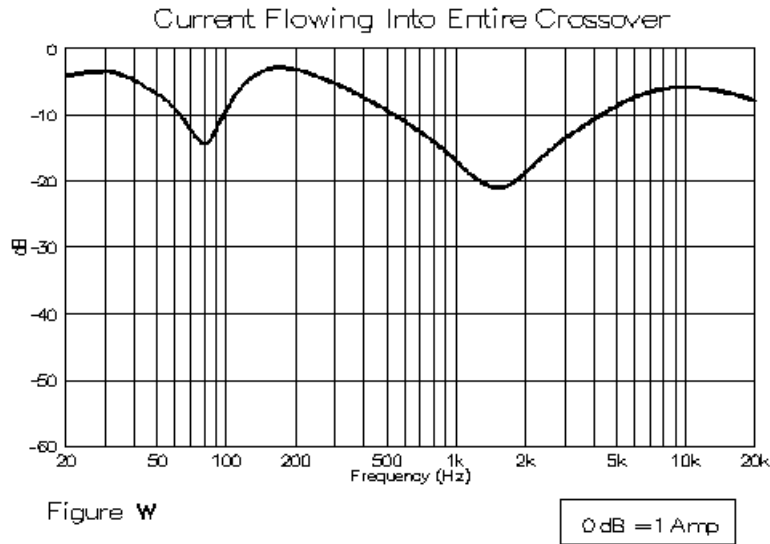
*
This is a different measurement than in my AES preprint, #4803, but it shows the same effect. - - - - The AES preprint outlines the development of the test signal, and shows measurements of many different audio components, including CD players, cassette decks, mini-disk players, loudspeakers, and digital electronics. - - - - The test signal can effectvely be used for any type of audio device except perhaps a microphone, and a loudspeaker with low enough distortion could measure a mic to the limits of it's residual distortion. **Current Draw of the Single Wire**
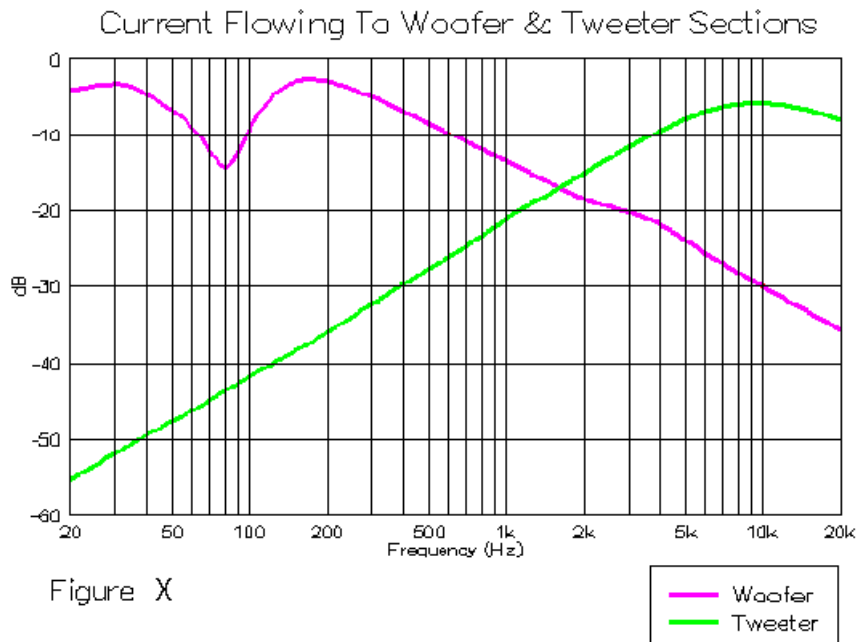The graphic below shows the current draw of the single speaker cable. - Compare this to Fig X on page7.

Current Flowing To Woofer & Tweeter Sections

Figure X

Woofer
Tweeter

Figure X
Current draw through bi-wire speaker cables

AES E-Library

A New Class of In-Band Multitone Test Signals

An in-band multitone test signal designed to maximize detection of harmonic, intermodulation, aharmonic, and cross-modulation products, despite the presence of multiple frequencies in the test signal, is presented. Measurements of electroacoustic systems are presented and contrasted against traditional harmonic distortion and intermodulation measurements. Measurements of other audio components, such as electronics, is also explored.

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AES Convention:105 (September 1998) Paper Number:4803
Subject:Instrumentation and Measurement

A NEW CLASS OF IN-BAND MULTITONE TEST SIGNALS
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Presented at the 105th Convention of the Audio Engineering Society
1998 September 26-29th, San Francisco, California as preprint #4803

Abstract

An in-band multitone test signal designed to maximize detection of harmonic, intermodulation, aharmonic and cross-modulation products, despite the presence of multiple frequencies in the test signal, is presented. Measurements of electroacoustic systems are presented and contrasted against traditional harmonic distortion and intermodulation measurements. Measurements of other audio components, such as electronics, is also explored.

0. Introduction

One of the stimuli for this project was a search for a test signal to better correlate with perceived listening clarity. Examination of a device using only harmonic distortion measurements does not provide a straightforward and consistent relationship between the measurements and listening results, especially once the measured harmonic distortion falls below a certain level. Traditionally used intermodulation test signals were considered, but these typically tend to provide only a few intermodulation or crossmodulation products within the audio band. Multitone signals looked to be the most promising type of measurement, as these would provide a complex stimulus for the DUT, and generate a larger number of intermodulation and crossmodulation products.

1.0 Limitations of Existing Signals

1.1 Harmonic Distortion

It can be confusing to try and correlate harmonic distortion measurements with what is heard. Once harmonic distortion levels are below several percent in the midband, it is a tossup as to which device will sound better or clearer. Up to a certain point, a given level of even order harmonics are more benign sonically than a given level of odd order harmonics, while higher orders are generally

more irritating than the lower order harmonics. A significant amount of second harmonic distortion may actually sound preferable to much lower amounts of third or fifth harmonics. A certain amount of second harmonic distortion may even be preferred to a system that has little or no second harmonic distortion. With all these confusing factors, harmonic distortion is not a clear cut way to evaluate a system or device once the levels have reached typical values for a modern high performance audio device.

1.2 Intermodulation Distortion

Intermodulation distortion measurements typically use just two tones, such as SMPTE, DIN, or CCIF, with a limited number of intermodulation and crossmodulation products within the audio band. There are a few three tone intermodulation tests that were proposed, but they have not become popular or mainstream tests. (1) The classic two-tone tests are severely limited in how much distortion across the audio band they will expose. Even performing tests using all the variations of the mainstream standards will still leave large gaps in the characterization of the DUT's intermodulation distortion performance, and consume a fair amount of time.

1.3 Spectral Contamination

Spectral Contamination testing has been proposed by Sokolich and Jensen (2). This looks more promising, but one of the proposed test signals covered in the paper is completely outside of the audio band, with signal stimulus extending from 150 kHz to 300 kHz, and it uses equal frequency spacing; in the example used in the paper, 9.36 kHz. The other proposed test signal spectrum extends from 10 kHz to 25 kHz, and has 120 Hz signal spacing. Equal frequency spacing means that many of the various intermodulation products and even some of the crossmodulation products will have the same resultant frequency. In the out of band version, this does generate a range of signals spaced 9.36 kHz apart in the audio band. For the 10 kHz to 25 kHz stimulus with 120 Hz spacing, the audio band distortion products tend to be at 120 Hz spacing, with a lot of the products at 120 Hz.

Placing the stimulus frequencies all outside of the audio band means that the device or system under test must possess greatly extended bandwidth and a concurrent extension of linearity, which may or may not relate to use within the audio band. It also means that the signal is ineffective for use with inherently bandwidth-limited systems such as loudspeakers, and most current digital audio systems. Additionally, the distortion products must all be of two kinds: crossmodulation products that divide down into the audio band and intermodulation products that subtract down into the audio band. No harmonics will be present within the audio band.

The version with the stimulus band from 10 kHz to 25 kHz still limits the use to wide band electronics, and precludes most digital and acoustic systems from being measured with this signal. It has similar restrictions as to what distortion products are displayed.

A third test signal is suggested, one which has signal components within the audio band, and a space or gap left in the middle of the band. This is designated as an "in-band" measurement. No frequencies or other details are suggested, and no measurements were performed using this signal in the Sokolich and Jensen paper.

1.4 SYSid Version of Spectral Contamination

Spectral contamination as performed by the SYSid measurement system has an in-band form of spectral contamination measurement. This signal has already found some use in correlating perceived clarity with the measurement (3). However, the usefulness is not as complete as it could be. In the default setup file, the frequencies either have equal spacing, or arbitrary spacing between the multiple frequencies. A total of 15 tones are used, with default frequencies of: 80, 100, 120, 160, 200, 240, 300, 400, 500, 700, 900, 1200, 1500, 2000 and 3000 Hz, which is spacing of: 20, 40, 60, 100, 200, 300, 500, 1000 Hz, this yields frequency to next frequency ratio's (or multiplier's) of 1.2, 1.25, 1.33, 1.4 and 1.5, not necessarily in any given order.

A formula is given in the owner's manual to achieve equal log spacing within a specified band, but this results in tone spacing that are a product of a given chosen bandwidth and number of tones, and has no other purpose other than to provide the equal spacing. The results from this formula are virtually random with respect to frequency spacing ratio's. A utility is provided called GENFN, which allows custom selection of the test frequencies for the SYSid spectral contamination test. The test frequencies selected and entered are placed into the closest FFT bin. This is done to maximize the ability of the FFT to examine the frequencies between these filled bins, but may make the spacing turn out to be less than optimum.

There is an admonition in the owner's manual to avoid making the test frequencies an integer multiple of one another, but this does not prevent the frequency spacing from generating intermodulation or crossmodulation products that can be covered up by a primary tone. For instance, intermodulation products that fall at 100 Hz or 200 Hz are completely masked by the primary tones at these frequency's. These frequency spacing allow the harmonics of the lower frequency test signals to be covered up as well, so even simple harmonic distortion products are hidden for some of the test signal tones.

The sheer number of tones can also present a problem in some cases, as the level of individual tones within the multitone signal must be at least 23.53 dB down from nominal in order to avoid clipping and other gross errors due to phase shifts in the system under test. This effectively limits the available measurement dynamic range with a 16 bit testing system to about 72 dB or less. The author is not aware of any capability within SYSid to set the relative phase of the various test tones.

Also, the number of tones within that frequency range makes it difficult to ascertain exactly where the distortion products are coming from, due to limitations in spectrum analyzer resolution. The density of the primary tones makes it hard to clearly ascertain the origin of the distortion products that are not already covered up by primary tones. This forces the use of the highest spectral resolution that the system is capable of, and still limits how accurately some of the distortion products can be ascertained.

Another concern with the SYSid default setup frequencies is that lower order harmonics are only present up to a certain point. The last primary frequency in the test signal is 3 kHz, and stopping the test signal content at this frequency limits the ability to determine both the level of harmonics for the rest of the audio band above 3 kHz, and the ability to detect intermodulation and crossmodulation products at higher frequencies. The limited band that the signal covers essentially limits it's use to the upper bass and midrange. Two or three way speaker systems would have virtually no excitation of the tweeter with this test signal.

The sequence starts out fairly low at 80 Hz, or even lower if a default file with a start frequency of 60 Hz is used, which is beyond the full capability of some smaller speaker systems and some other bandlimited systems. In my opinion, a widely acceptable test signal should be useful on almost any audio device, even a bandlimited one. A low frequency of 100 Hz is more appropriate for a test signal to have a more universal application.

To the credit of the designers, the flexibility of the SYSid test system allows the user to generate a user defined set of spectral contamination test tones. This will enable the SYSid system to take advantage of the newly designed test signals presented herein.

1.5 Audio Precision Multitone Signals

The Audio Precision System One DSP and System Two both have a multitone in-band test signal capability, with the default setup having frequencies spaced according to 1/3 octave ISO centers, which is a multiplier ratio of 1.25. With this spacing, every third frequency is spaced an octave apart, and the resultant frequencies often end up at even spacing amounts. A total of 31 separate tones are used, which reduces the dynamic range available. Individual tones must be at -29.83 dB from nominal, unless phase manipulation is used to reduce the crest factor. As noted earlier, this may not be a good idea if the system or device under test is known to have amplitude roll-off in the audio band, excess phase shifts or time delays built into the system.

Audio Precision does adjust the frequencies to correspond to an FFT analyzer bin center frequency, in order to maximize the ability to use the empty analyzer bins to best advantage. These 31 frequencies are: 16.15, 21.53, 26.92, 43.07, 53.83, 64.60, 80.75, 102.28, 123.82, 156.12, 199.18, 253.02, 317.61, 398.36, 500.65, 635.23, 802.11, 1001.3, 1248.9, 1598.8, 1997.2, 2503.2, 3154.6, 3999.8, 4995.7, 6352.3, 7999.6, 10002, 12500, 16005, and 19999 Hz. Note that none deviate from the standard ISO centers by more than about 5 Hz.

This does offset the exact regularity that would tend to occur with such ISO-center frequency spacing, but since the analyzer bins are limiting the ability to resolve the distortion products to any better accuracy, these minor offsets are not very effective in avoiding intermodulation and crossmodulation product cover up, or harmonic cover up by the primary tones. Even with the slight offset in the stacking up of harmonics, intermodulation and crossmodulation products will not be able to be resolved due to the inherent width of the FFT bins.

The Audio Precision systems can be programmed or set to generate and test using any frequencies chosen by the user, but they do not offer a means or advice on avoiding the generation of multiple distortion products at the same resultant frequency, and at the same frequencies as some of the primary tones.

There is an AP codec test signal (5), which has two bands of 8 frequencies (16 total) separated by a two octave gap. These are not all at ISO centers, +/- the FFT bin center, but they still exhibit some stacking of distortion products. This test signal is similar to the one proposed by Sokolich and Jensen.

The frequencies are: 53.83, 123.82, 209.95. 312.23, 446.81, 608.31, 807.50, 1060.5, 3999.8, 5022.6, 6287.7, 7859.6, 9808.4, 12230, 15234, and 18965 Hz.

The frequency spacing for the various tones is not according to any pattern or design that is readily apparent. Information about this particular AP test signal was not received until the basic research for this paper had been completed.

2.0 New Spectral Contamination Signal

After discovering that harmonic, intermodulation, and crossmodulation products were being masked to one extent or another by the present implementations of spectral contamination tests, I sought a series of frequencies which would avoid this problem. A number of options were explored, and while frequencies could be picked at random, or arbitrarily, it was felt that some sort of defining function or multiplier would be more useful and consistent. It was found that a multiplier based on the Golden Ratio, or the final ratio of the mathematical Fibonacci sequence, of 0.618034 worked very well. By using a multiplier of 1.618 or 2.618, a sequence of tones is generated that avoids any of the harmonics, and almost all of the intermodulation and cross-modulation products from being covered up spectrally by any of the other tones or products.

2.1 Working Criteria

It was desired that the tone sequence start and stay within the audio band, so that bandlimited systems would be fully excited by the test stimulus. Accordingly, 100 Hz was arbitrarily chosen as a start frequency that was well within the range of even small multimedia loudspeakers. This results in a series of 6 tones using the 2.618 multiplier at frequencies of: 100, 261.8, 685.4, 1794.4, 4697.9, and 12299 Hz for the test signal I call the Phi6 spectral signal. Figure 1 graphically depicts the FFT spectrum of the signal as output from a CD player, through a mixer for level control.

In an effort to limit the loss of dynamic range, it was decided that there should be a practical limit to how many tones were used. Starting from 100 Hz, and using the 1.618 multiplier, 12 tones within the audio band result, and this was felt to be a good upper limit.
Using a much lower frequency only results in a few more tones within the audio band, and these frequencies are all fairly low.

The 12 tone signal using the 1.618 multiplier results in the former frequencies plus: 161.8 Hz, 423.6 Hz, 1109.0 Hz, 2903.4 Hz, 7601.3 Hz, 19900 Hz, and is called a Phi12 spectral signal. Figure 2 graphically depicts the FFT spectrum of the signal from the CD player, and through the mixer.

Further refinements and variations on the test signals suggested themselves after working with the proposed test signals to actually measure a system. These variations are covered in the experimental results section, as they were developed as a result of the experimentation.

3.0 Experimental Results

Experimental multitone test signals were generated using a PC based waveform generator operating in the digital domain, and transferring the final product to CD-R for playback and use.

It was found that some CD players were unable to handle output to full 0 dBFS, with the result a much higher level of distortion. It was as if some component was either clipping or compressing the signal, generating harmonics, intermodulation and crossmodulation products all throughout the spectrum. Accordingly, it was found that reducing the level of the digitally generated recorded signal by 2 dB was enough for most players to avoid this problem. In order to avoid any vestige of amplitude related compression or clipping, the signals were generated and recorded with peak signal levels below -3 dB. Some players still had a problem with these peak levels, but most did not. The author suspect's the digital filtering within the CD player may be the cause.

3.1 Revisions

The original Phi12 stimulus had some distortion product cover up, due to some of the intermediate tones intermodulating with each other, and generating the same frequency. Once this was discovered, a revised version that used different multipliers for each of the 6 base tones was devised. While this did include some arbitrary multiplier choices, and could conceivably be improved, it was found useful as a more complex stimulus than the 6 tone version.

Accordingly, from the base frequency of 100 Hz, a multiplier of 1.22 was used, resulting in 122 Hz. From 261.8 Hz, a multiplier of 1.33 was used. resulting in 348.2 Hz, from 685.4 Hz, a multiplier of 1.44 resulting 987.0 Hz, from 1794 Hz a multiplier of 1.6 resulting in 2870.4 Hz, from 4697.9 Hz, a multiplier of 1.44 resulting in 6765.0 Hz, from 12299 Hz, a multiplier of 1.33, resulting in 16358 Hz. None of these has obvious intermodulation or crossmodulation products that occur at exactly the same frequency. Most distortion products are far enough apart for a spectrum analyzer to differentiate between the different products. Figure 3 is the FFT spectrum of this signal, from a CD player through a mixer.

4.0 Further Test Signal Refinements

Other changes and refinements were considered to reduce or alleviate some of the limitations of the signal.

4.1 Possible Crest Factor Reduction

Proper phasing of the various signal tones can provide some relief, but a DUT that had significant amplitude changes, roll-off or phase shifts would destroy the carefully orchestrated phase relationships of the original test signal. It is recommended that full allowance be made for the worst case crest factor of the multiple tones. Several references regarding such phase manipulation are provided. (4)

4.2 Split Octave Bands - High Low

After working experimentally with some of the variations on the signals, it was determined that more tones or a higher density of tone spacing might be desirable to highlight intermodulation and cross-modulation distortions, and provide more distortion products across the audio band. It was also determined that limiting the total number of individual tones would help preserve dynamic range.

To raise the tone interval density without increasing the number of tones to a high number, isolated octave bands of tones was considered. A suitably non-overlaying multiplier was desired, so a multiplier of one plus one tenth the original full audio band multipliers were investigated. Scatter diagrams were used to evaluate the suitability of various octave band combinations with a reasonable total number of tones, that would provide a wide range of harmonic, intermodulation, and crossmodulation products. One of the concerns was to limit the spread of each group or band of signals to an octave or less, so that the second harmonics of the band would not be covered up by the primary tones.

A pair of split octave bands was found to provide an efficient and useful multitone test signal without an excessive number of individual tones. An octave starting at 100 Hz, in conjunction with an octave starting at 5 kHz, with both containing five tones, and with spacing multipliers of 1.1618. This results in frequencies of 100, 116.18, 134.98, 156.80 and 182.19 Hz for the low band, and of 5000, 5809, 6748.9, 7840.9, and 9109.6 Hz. This test signal proved particularly efficacious at ferreting out intermodulation and crossmodulation in two-way loudspeakers. Figure 4 depicts the FFT spectrum of this signal, from a CD player through a mixer.

Note that this is similar in spirit to the third signal proposed by Sokolich and Jensen, but not identical. Much larger areas are left open for distortion products to manifest and be readily observed.

Other split band combinations were investigated, such as:

100 -182 Hz plus 10 - 18.2 kHz, with frequencies of 100, 116.18, 134.98, 156.80 and 182.19 Hz for the low band, and 10000, 11618, 13498, 15682, and 18219 Hz for the high band. Figure 5.

100 -182 Hz plus 1.11 - 2.02 kHz, with frequencies of 100, 116.18, 134.98, 156.80 and 182.19 Hz for the low band, and 1109.0, 1288.4, 1496.9, 1739.1, 2020.5 Hz for the high band. Figure 6.

261 - 477 Hz plus 1.11 - 2.02 kHz, with frequencies of 261.80, 304.16, 353.38, 410.56, and 476.99 Hz for the low band, and 1109.0, 1288.4, 1496.9, 1739.1, 2020.5 Hz for the high band. Figure 7.

261 - 477 Hz plus 10 - 18.2 kHz, with frequencies of 261.80, 304.16, 353.38, 410.56, and 476.99 Hz for the low band, and 10000, 11618, 13498, 15682, and 18219 Hz for the high band. Figure 8.

685 - 1249 Hz plus 1.79 - 3.27 kHz, with frequencies of 685.41, 796.31, 925.16, 1074.9, and 1248.8 Hz for the low band, and 1794.4, 2084.8, 2422.1, 2814.0, and 3269.3 Hz for the high band. Figure 9.

 4.3 Revised Split Octave Band Signal

It proves all too easy to fall into the same trap as before. In selecting even starting frequencies, such as 100 Hz for the low band, and 10 kHz for the high band, some of the harmonics, intermodulation and crossmodulation products line up again, covering up the source of the tones responsible for the distortion.

After trying a different start frequency that was not an even multiple of the other start frequency for the high band or the low band, it was realized that by using the same multiplier for each band, the top band of frequencies would divide into the bottom band and have the same crossmodulation product. This leads to the need for a different multiplier for the separate bands. After checking for this, a slightly different multiplier for one of the bands was used, 1.2618 vs. 1.1618.

Use of the new 1.2618 multiplier results in the high frequency band extending for a bit more than an octave, but the author considers this to be completely arbitrary and a minor issue. If a band of closely spaced multitone frequencies extends past an octave, some potential exists for cover-up of harmonics. For the higher frequency bands where this was done, the resolution is sufficient to allow determination of the 2nd harmonics without difficulty. This is why the low frequency band uses the originally derived multiplier.

At this point, a spread sheet was developed for use in evaluating the resultant distortion products that would occur with various primary test signal tones in a spectral type test signal.

Figure 10 is a copy of the spread sheet for the Phi6 spectral signal, Figure 11 is a scatter diagram of the primary tones for the Phi6 spectral, harmonics (up to the tenth), and intermodulation and crossmodulation products. Figure 12 is the spread sheet for the Phi12 spectral signal, and Figure 13 the scatter diagram. Figure 14 is the spread sheet for the Phi12r (revised), and Figure 15 the scatter diagram. Figure 16 is the spread sheet for the first split band spectral, the one with bands at 100 -182 Hz plus 5 - 9.1 kHz.

Experimental results for the first set of split band spectral test signals reveal that the pairings with the low band at 261 to 477 Hz did not show any additional distortion products than the ones with the low band from 100 to 182 Hz. They also do not show low frequency second harmonics as readily. Also, the pairings that had the high band in the last octave do not show any high frequency second harmonics, or intermodulation or crossmodulation products on the high side before the audio band was exceeded. In order to address this, revised split band signals were reduced to a set of low and high band signals, and a set of low and middle band signals.

The revised split band spectral's now use frequencies at: 100 -182 Hz plus 4.7 - 11.9 kHz, with frequencies of 100, 116.18, 134.98, 156.80 and 182.19 Hz for the low band, and 4697.9, 5927.8, 7479.7, 9437.9 and 11909 Hz for the high band for the low/high band pairing, and called a Phi Low-High split band spectral. Shown in Figure 17.

The low/middle revised split band now uses frequencies at: 100, 116.18, 134.98, 156.80 and 182.19 Hz for the low band, and 986.99, 1245.4, 1571.4, 1982.8 and 2502.0 Hz, and I call it a Phi Low-Middle split band spectral. Figure 18.

4.4 Phi Tri-Band Spectral Signal

Further work with these split band spectral signals resulted in the development of a split band signal with components at low, middle AND higher frequencies, which left two large open areas for distortion products to show up. The multipliers have to be different for each band. The total number of tones needs to be kept down, so three clusters of four frequencies was used, for a total of 12 separate tones. Multipliers of 1.1618, 1.2618 and 1.12618 were used for the three bands respectively. The frequencies that result are: 100, 116.18, 134.98, and 156.80 Hz for the low frequency band, 986.99, 1245.4, 1571.4, and 1982.8 Hz for the middle band, and 6764.9, 7618.5, 8579.8 and 9662.5 Hz for the high frequency band. I refer to this one as a Phi tri-band spectral, shown in Figure 19.

The frequency ranges were chosen based on earlier work with split band multitone's, and what the author felt would be good ranges for excitation frequencies. Each band has an empty space of approximately two octaves in-between, the low band two octaves below, and the high band an octave above (within the audio band). The start frequency of each band is based on the revised Phi12 (Phi12r) spectral contamination test signal. The underlying premise is that none of the frequencies within each band, including the start frequency, would encourage distortion products to be covered up by other products, or by the primary tones.

5.0 Actual Measurement Results

Unless otherwise noted, the electrical tests all had their level adjusted with a mixer to bring the signal up to the same approximate input level to the FFT spectrum analyzer, so as to maximize dynamic range and the display of potential distortion products. The mixer and complete signal playback system used as the source for the tests were checked to assure that the distortion components arising from these components did not add to, or limit the measured performance of any of the tests.

 5.1 Electrical, Record/Playback Devices Measured

Several CD players were checked with the new signals, the professional balanced output player used as the signal source worked well, and a top of the line consumer CD player had an even lower noise and distortion floor, see Figure 20. A portable CD player with a line out showed good results too, see Figure 21, the gray spectrum.

Headphone output jacks on portables were another story, as depicted in the black graph of Figure 21. This is the same player, the lower curve out of the line level jack, and the curve showing a higher level of distortion and noise being from the headphone out jack.

Several CD players measured had problems when the recorded signal had peaks at digital 0 dBFS, where they exhibited symptoms of clipping/compression. Figure 22 shows this effect to a moderate degree. Figure 23 depicts the difference between 0 dBFS recorded levels, and -6 dBFS recorded levels on another portable player. It can be seen that a reduction of recorded level by 6 dB has reduced distortion by over 20 dB. Gain has been added via a mixer to keep the input levels to the FFT analyzer the same, the only difference is the level recorded on the CD.

A high quality consumer cassette deck was examined via a record/playback of the test signal, using Dolby B noise reduction. The source was the high quality CD player shown in Figure 20. Figures 24, 25, 26 and 27 show the response of the cassette deck when the test signals were recorded at 0 VU levels, accounting for pre-emphasis. Note the presence of distortion and modulation components at fairly high levels. The high frequency content of the test signals in Figures 24, 25 and 27 prevents the normal advantage the Dolby B high frequency noise reduction provides.

A consumer Mini Disc deck was tested, and the results depicted in Figures 28, 29, 30 and 31. In Figure 28, the Phi12r spectral signal, the noise and distortion in-between the signals reaches levels as high as with the cassette deck. Note the generally lower levels of distortion products and noise in between the signal bands compared to the cassette deck with the split band signals. One of the original split band spectral test signals did show a high frequency roll-off with the band containing frequencies at 100 -182 Hz plus 10 - 18.2 kHz. The last tone in the high frequency band was noticeably attenuated, in contrast to the high quality consumer cassette deck. See Figure 32.

A digitally based crossover and filter system was tested using the Phi12r spectral test signal, shown in Figure 33. The test signal passed through the A to D and the D to A stages without any digital filtering or EQ operations being performed on the signal. Note the rise in the noise floor of over 10 dB, and a few very low level products, compared to the source CD player. Interestingly enough, once a filtering algorithm was engaged, the noise floor went down some, as seen in Figure 34, yet it is still higher than the source.

As can be seen from the test measurements, the new signal is effective in testing electronics, digital systems, and recording systems. I have no doubt that it would be useful for the testing of various transmission systems as well, especially transmission systems using companding. However, access to such systems was limited, and the author was unable to present measurement results within this paper.

5.2 Electroacoustic Loudspeaker Tests

The new test signal proved highly effective in measuring loudspeakers, as they tend to have a much higher level of intermodulation and crossmodulation than electronics and similar systems. Changes in the distortion content with changes in drive level were easily noted. All loudspeaker measurements were taken in an anechoic chamber, at 1M, unless otherwise noted.

Figure 35 shows the distortion for a commercial plastic injection molded 15" based two-way system with a 1" throat compression driver (called Speaker A), at one watt RMS drive levels for the Phi6 test signal. Note the distortion products clearly visible in the middle/high range. Figure 36 is this same speaker at 10 watts RMS drive level, and the distortion has risen considerably. Figure 37 depicts this same system with the Phi12r test signal. The higher density of potential distortion products allows a clearer picture of where the speaker is running into trouble. Contrast this with the overlay of the same loudspeaker tested with all the different original split band spectral signals that were explored (Figures 4 through 9) , in Figure 38. An obvious problem can be seen centered around 1 kHz, and again at 4 kHz. With only one measurement, the Phi12r reveals substantially the same clues of trouble. The tri-band spectral also provides a similar level of information with one measurement.

Conventional harmonic distortion plots do not provide much of an indication that this speaker has a problem. See Figure 39. The levels of harmonic distortion are not unusually high, and do not predict that this speaker will sound fuzzy or less clear than another. Further spectral signal test measurements at 10 watts are shown in Figures 40, 41, and 42.

To provide a reference point, a prototype plastic injection molded 15" based two way with a 1" compression driver (Speaker B) is compared. In Figure 43, it can be seen that speaker B has 10, and up to 15 dB less distortion than speaker A. The harmonic distortion plots for speaker B are not radically different than those for speaker A, yet the intermodulation and crossmodulation is substantially lower. Figures 44, 45, and 46 show the comparison with the split band spectral signals.

A 2-way nearfield studio monitor prototype using a 6 1/2" woofer, and a 1" titanium dome tweeter is tested at 1W RMS in Figures 47, 48, and 49. The speakers are measured at 1/2M to increase the signal to noise ratio, and attempt to show the very low spectral contamination of the unit. The input level is consistent with intended usage, and it should be noted that a 1W RMS signal level has peaks of close to 100W with these multitone spectral signals! This is excellent performance for a loudspeaker system by any standard, and provides an interesting contrast with the sound reinforcement speakers measured above (Speakers A and B).

Individual drivers can be tested using these signals. An inexpensive stamped frame woofer is tested in Figures 50 and 51. The test signal is input to the woofer full range. Measurements of a compression driver tweeter in Figures 52 and 53 are filtered at 1 kHz at a 12 dB/octave rate to protect the driver, but the low frequency components of the signal still have their effect on the total distortion, and increase the presence of
low frequency intermodulation products.

5.3 Electrical Current Tests on Bi-Wired Speaker Cables

One of the other original motivations for investigating this type of test signal was the desire to be able to objectively explore the performance of the various forms of audio cables: line level interconnects, speaker cables, microphone cables, etc. In that respect, it is a bit of a disappointment, but one interesting measurement result did develop. For many years, a debate has raged regarding any possible benefit of using two separate cables to wire a speaker with electrically separated crossover sections, or bi-wiring as it is known.

Theory indicates that in any advantage is to be had, it would have to manifest as a reduction in intermodulation between the low frequency currents and the high frequency currents in the cable. Separating the two ranges should provide a measurable benefit.
The signals developed and refined in this paper are highly sensitive to intermodulation and crossmodulation distortion products, so what better test signal than a multitone spectral?

A wide-band current probe was used to measure the current flowing through the speaker cable/s. First the single speaker wire was measured, Figure 55. These are distortion products of the electrical current signal through the cable, and can be seen to be about 45 to 55 dB down from the primary tone's level's. Then the current transformer was placed on the tweeter cable of a bi-wiring arrangement, Figure 56. Note the reduction of low frequency currents in the first spectral tone band, and the reduction of distortion products by 20 dB or more through the entire midrange. Figure 57 depicts the current flowing through the woofer side of the bi-wire cables. Some reduction in distortion products is present in the high frequency regions, starting as low as 1.5 kHz, and reaching reductions of 10 dB. Figure 58 is the full range single cable overlaid with the tweeter bi-wire cable, and the reduction in distortion products is readily discernible.

Tests on different single speaker cables have so far been inconclusive, with inconsistent differences of only 2-4 dB at the limits of the test system resolution, which are not definitive enough to be able to draw any conclusions about. Interesting measurement and data, nonetheless.

6.0 Conclusion

An improved method of selecting frequencies for a multitone test signal has been presented , one which allows the distortion products from the test signal to avoid being covered up by the primary signal tones or other distortion products. Measurements taken using the improved method have indicated that it is a highly effective test signal, and shows promise for use in testing not only electroacoustic systems, but audio electronic, recording and transmission systems. Standardization and selection of such a multitone signal could prove useful to all audio engineers for use with virtually all aspects of audio system and component testing, and may lead to a better correlation of objective measurements with subjective listening test results.

7.0 Acknowledgements

8.0 References

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(2) Deane Jensen and Gary Sokolich, "Spectral Contamination Measurement", AES Preprint #2725

(3) Mix magazine, May 1998, p. 142, review of the Benson StudioStat 8.2, Lab analysis, by Mike Klasco and June 1998, p. 174, review of the KRK Systems Expose Series E-8, Lab Analysis by Jack Hidley.

(4) Stephen Boyd, "Multitone Signals with Low Crest Factor", IEEE Trans. on Circuits and Systems, Vol. 33, #19, Oct. 1986
Mathias Friese, "Multitone Signals with Low Crest Factor", IEEE Trans. on Communications, Vol. 43, #10, Oct. 1997
Larry J. Greenstein & Patrick J. Fitzgerald, "Phasing Multitone Signals to Minimize Peak Factors", IEEE Trans. on Communications, Vol. 29, #7, July 1981

(5) Richard C. Cabot, "Performance Assessment of Reduced Bit Rate Codecs", presented at the AES Conference "Managing the Bit Budget", London, May 1994. Available as a reprint from Audio Precision upon request.

Appendix A - Equipment used in analysis of spectral test signals

CD player, primary signal source from digitally generated CD-R's: Marantz PMD 321
Mixer: Peavey RSM 1662
Microphone: ACO 7012
Mic Preamp/Meter: Larson Davis Model 800B
FFT Analysis, HD plots, Frequency Response Plots: TEF 20, SLX software, FFT 8192 point mode
Power Amplifier: Peavey CS-1200
Wide-band Current Transformer: Pearson Electronics Model 411

Bi-wiring

You might want to try bi-wiring, but it helps to use something better than just zip cord, no matter what the gauge.

If you are running 10-12 foot lengths, a decent single wire set of speaker cables will run around $140 or so, so double that for bi-wiring the two front mains. I would not worry about bi-wiring the center speaker, or using anything but 12 gauge zip for the surrounds, as the suround signal is processed through a cheap digital delay as well as derived from signals buried in the main signal, and hence is not that clean or wide bandwidth.

See this post for a list of decent aftermarket cables that don't cost a fortune, including references to DIY cables:

http://www.audioreview.com/message/cgi-bin/dcforum/dcboard.cgi?
az=show_thread&omm=0&om=809&forum=DCForumID8

How To Hook-Up Bi-Wiring
See: http://www.soundstage.com/synergize/synergize031998.htm
AND/OR
http://www.davidmannaudio.com/faq/faq3.html

and if that is not detailed enough:

For each speaker, instead of hooking up a single cable to the amp channel, and then connecting it to the jumpered (+) and (-) at the speaker, you REMOVE the jumper bar from between the (+) and (+) terminals top and bottom, and from between the (-) and (-) terminals top and bottom. Then you determine which pair (top or bottom) is the pair for the tweeter. Usually the top pair is for the tweeter, but check your owners manual to be sure. For the sake of discussion and detailing tthe hookup, let's assume that the top pair IS for the tweeter. If this is not the case, then modify your instructions accordingly.

Now, you hook up the (+) and the (-) to the amp output terminals for that channel, connecting BOTH speaker cables at once. Specifically, connect the (+) of the woofer cable AND of the Tweeter cable to the (+) terminal of the amp. Using spades, this can be done relatively easily by stacking the two spades on top of one another and tightening the amp post down. It would help to have a helper hold the two spades firmly in place while you tightened the post. After you are done, do not tug on this connection too much so as to avoid loosening it.

Alternatively, if either cable has a banana plug, then if only one has a banana, then connect one via spade, and plug the banana in after that. If both have banana plugs, then stack the banana plugs up into the amp terminal. I prefer dual spades, or a spade and a banana, but two banana's get's to be a somewhat poor connection, and even if you can place the woofer cable's banana first, it is still not optimum. Then connect the other polarity (-) of both cables to the (-) of the amp, as appropriate for the connectors used.

Now connect the woofer cable to the bottom pair (woofer) of speaker terminals, keeping the (+) and (-) correct, and connect the tweeter cable to the top pair (tweeter) keeping the (+) and (-) correct. Make sure all connections are tight, and that none of the terminals is shorting to any metal or the other terminal.

You have now hooked your speakers up in bi-wire mode.

I would expect that you would hear smoother bass, a clearer midrange, and more detailed yet smoother and clearer highs after this hookup compared to the old single cable run.

How Bi-wiring Works

Bi-wiring is accomplished via separate pairs of terminals on the loudspeaker system, typically one pair for the woofer, and one pair for the tweeter or midrange and tweeter. They are completely separated electrically from one another. The normal function of a loudspeaker crossover is to guide the proper frequency's to the proper driver. Lows to the woofer, and highs to the tweeter. This is done in part for protection from the division of labor that has occured with two disparate speakers: tweeters will be damaged or destroyed if exposed to low frequency's and woofers just heat up when exposed to the higher frequency's, as they are too massive to respond at all. The other function that a crossover provides is in allowing the two speakers to blend together, to mesh with one another to become a single apparent sound source. They can also provide some passive EQ of the drive units, as long as there is excess energy to throw away.

The fundamental way a loudspeaker crossover works is to vary the impedance seen by the speaker and by the power amplifier. In the case of the woofer, the crossover network for it has a very low series impedance at low frequency's that gets gradually higher and higher in impedance between the amp and the speaker at higher frequency's. For very low frequency's, there is lots of current flow to the woofer, and for higher frequency's, there is little current flow due to the much higher impedance. In the case of the tweeter, at low frequency's the series impedance is very high and very little current flows, and as the frequency goes higher, the impedance of the crossover network gets lower and lets through more current.

The situation is such that when the full range musical signal is applied to the terminals of a full-range speaker system, the woofer only gets sent low frequency signals, and the tweeter only gets sent high frequency signals. Once the crossover networks have been electrically separated, they still continue to function in the same manner, having a low impedance in their passband of application. This means that if separate speaker cables are hooked up for the woofer and it's portion of the network, and the tweeter, and it's portion of the network, not only have the speakers and the frequency's directed and divided for them, but the two separate speaker cables will now also carry different signals, the woofer cable mostly the lows, and the tweeter cable mostly the highs.

Once the highs and lows have been separated in this fashion, the strong current pulses and surges that a woofer demands when reproducing bass or drums will not interact with the delicate sounds of a flute or cymbal. The magnetic field of the low frequency signals cannot modulate or interfere with the highs, and to a lesser extent, the reverse is true.

Now that the low and high frequency signals have been divided among not only the speaker drivers, but the speaker cables, these cables can be more specialized for their intended purpose. The woofer cable can concentrate on low DCR, and not have any big concern for extremely low inductance, the tweeter cable can be designed for very low inductance, and not as concerned about total DCR.

Using one much larger unsophisticated cable to achieve the same thing as bi-wiring is just not possible, the separation of work has not occured, and the ability to optimize each separate run is not available. Additionally, as the gauge of a wire decreases (wire gets fatter) and the spacing between the pair of wires that constitute a speaker cable gets greater, the inductance tends to go up. Using one larger unsophisticated cable actually makes things worse for the tweeter, as even though the DCR has gone down and the woofer gets more energy compared to the thinner single cable, the tweeter now gets less energy in the extreme highs. The net result is a shift in the tonal balance that can even exceed the criteria held dear by the ABXers.

The current path's I describe can easily be plotted, measured and verified by any speaker or cable engineer. There is absolutely no doubt whatsoever about their existence or validity. In point of fact, properly implemented bi-wiring has benefits that can not be achieved by a single unsophisticated cable or even a single exotic cable.

Jon Risch

Bi-Wiring 101

In order to explain how bi-wiring works, it is necessary to explain a bit about how crossovers work. It will also be necessary to contemplate more than the usual voltage output of the crossover sections, so do not assume that if you know the basics for crossovers, that you will know what this will be all about.

Let's look at a simple two-way system with a first order crossover, the simplest crossover and system we can examine. It will be relevant to other more complex systems, so once you understand this one, the others will fall into place. We will not address the issues of tweeter level padding, response EQ, etc., just the basic crossover function itself.

In a simple first order crossover, there is an inductor in series with the woofer, and a capacitor in series with the tweeter. These two components comprise the crossover system. Normally, these two components are connected to the same input terminals on the speaker, in parallel. Hence this type of crossover topology is called a parallel type crossover.

A full range voltage signal is sent down a speaker cable, and appears at the single pair of input terminals. A current is drawn based on the input impedance of the speaker system as a whole, which in most cases, will have a relatively flat impedance curve once we get above the bass resonance region, where the impedance will be dominated by the cabinet design resonance's. If we say that (for purposes of this discussion) the overall impedance of the speaker in the midrange and on up is relatively flat, then a consistent amount of current will flow through the single speaker cable all across the audio band.

So there are several elements to the total circuit formed by the amp output terminals, the speaker cable, and the speaker system and crossover network. A signal appears at the amp terminals, represented by a voltage, the impedance of the speaker system causes it to draw an amount of current proportional to it's impedance for a given drive voltage, and this current flows through the speaker cable.

Now in order to examine what happens when we bi-wire, it will be necessary to go into some of the detail as to how a crossover "crosses over". If we look at just the woofer, and it's series inductor, the inductor provides little impediment to low frequencies traveling through the inductor, and a high amount of impediment to the higher frequencies. Looked at another way, the inductor impedes the highs but not the lows. If we examine an impedance curve of just the woofer with its series inductor, we would see that the impedance was pretty much just that of the woofer in the low frequencies, and would rise with frequency as the inductor impeded more and more of the highs. For this situation with just the woofer, for a given voltage drive level, a certain amount of current would be drawn at low frequencies, and this amount would decrease as the frequency went up, due to the rising impedance.

If all that was hooked up to the amp was the woofer and it's associated inductor, then the current flow in the single speaker cable would follow the impedance curve, a certain amount of current flow at low frequencies, tapering off at higher frequencies. Perhaps a glimmer of the true situation with bi-wiring is beginning to appear.

Now let's just look at the tweeter, and it's associated capacitor in series. At low frequencies, a capacitor tends to impede the flow of current, and at high frequencies, it provides little impediment. Hence, when we hook up just the tweeter and it's capacitor to the amp terminals through the single speaker cable, there is little current flow at low frequencies, and an increasing amount as the frequency goes up. At some higher frequency, the current draw is determined by the impedance of the tweeter alone.

Now just to make sure that it is understood, it is the current flow through a dynamic driver (one with a magnet and a voice coil) that causes it to move. A voltage applied that had no current capability would not cause any movement. This means that in order for the voltage at the amp terminals to cause a speaker to move, it must have a relatively low source impedance, so that when a given voltage appears at the amp output terminals, a given amount of current can flow into the load's impedance. That is why when the crossover components impede the current flow, they cause the output of the driver to drop off, hence the crossover function is achieved.

Note that the woofer and it's associated inductor, and the tweeter and it's associated capacitor will function independently, they roll-off the frequencies out of the driver's operating band without regard to whether or not the other half of the crossover is present or not. When both sections of the crossover are present, and connected in parallel, the overall impedance curve looks relatively flat, as when the tweeter section has it's impedance going up in the low frequencies, the woofer has it's impedance going down. At the crossover point they are more or less equal, and this is the point in frequency at which the impedance's of the two sections in parallel equal approximately half that of either section alone. This is how two 8 ohm drivers can be connected together through a crossover, and not equal a total load of 4 ohms.

By now, you should be getting the idea about bi-wiring. Instead of one speaker cable, or just one of the drivers and it's associated crossover component being connected to the amp's output terminals, two separate speaker cables are connected to the same amp output terminals and run to the now separated crossover sections. With different impedance's being presented across the audio band, each cable carries a different signal than a single speaker cable. The separate cable for the woofer carries mostly the LF currents, and the separate cable for the tweeter carries mostly the HF currents. This is due to the differing impedance's we discussed above.

Now if all you think of is the voltage at the amp terminals, and how the two cables are carrying the same voltage to the woofer and the tweeter sections, then it still may seem that the same signal is being delivered to the drivers as through one speaker cable. IF the speaker cables were perfect, and had zero impedance, infinite mass, and no digressions from ideal LCR behavior (DA, DF, hysterisis, etc.), then it may be that this would be the case. Since the cables we have available to us are not perfect, there are losses in the cables.

The $64,000,000 question is, how much does the real world speaker cable compromise the performance of a speaker compared to bi-wiring?

To make this easy to figure out, we will ignore the effects of inductance and secondary effects, and focus strictly on DCR effects. Let us assume that a cable sufficiently large enough to keep speaker system impedance variations from affecting the amplitude response by more than 0.1 dB was used, meeting the Krueger criteria. In many cases, this is a very large cable, usually at least a 14 gauge, and often 12 gauge OR LARGER.
For a copy of the Krueger criteria see: http://x42.deja.com/[ST_rn=ps]/getdoc.xp?AN=450322078&CONTEXT=927059192.1901920287&hitnum=6
(I should warn that I do not agree entirely with Arny's criteria, it completely ignores inductance, which typically gets worse as a ratio of DCR to HF impedance of the cable as the gauge gets smaller, or larger zip cords)

How quick do the gauge requirements add up? If you have only 10 foot cables, and a speaker with a minimum Z of 6 ohms, then a 14 gauge wire is necessary to prevent any more than a 0.1 dB amplitude variation due to the cable DCR. If the speaker Z minimum hits 3.7 ohms, you are now up to 12 gauge. Anything longer in terms of the speaker cable, or lower in terms of the Z, will require larger than 12 gauge to reduce the amplitude variations due to voltage drops to less than 0.1 dB.

See:

[Bi-Wiring Attenuation of Inter-Driver IM](#)

If we were to look at the simple change in DCR from merely doubling up on the cable, then distortion would only go down 6 dB, from halving the DCR and nothing more.

Of course, once we start using real music, with more than just two frequencies, and real world cable situations that might have even more DCR, and the inductance differences between a single zip cord and two high performance speaker cables, the amount of distortions in a single speaker cable go up considerably, and the amount of reduction in distortion is increased for the bi-wire comparison. This means that we might be into 2% IM or more, and with multiple frequencies, which make it even worse sounding.

All of the above totally ignores any potential magnetic field interactions, many of which would be time delayed and would smear out transients and large signals. The magnetic field distortion reduction would come from the separation of the LF currents and the HF currents.
The time delayed and resonance associated signature would tend to make these distortions even more noticeable than the self-IM of the cable due to voltage drop.

I think it is easy to see that a multidriver system with higher order crossovers will react similarly to this very simple first order two-way system that has been analyzed.

It is interesting to note that higher order crossovers tend to have a similar input impedance for each section as a first order, and it is the output signal of the various sections of the crossover that are made to roll off steeper. In essence, the reductions in current for each cable in a bi-wire pair will be at a 6 dB/octave slope almost regardless of the crossover order.

Jon Risch

Last ditch explanation for those who still don't get it.

Assuming that you have read the above information, and have the beginings of an understanding of how a crossover works, and how it divides the frequencies to the speakers, lets try this:

Traditionally, a crossover cirucit for the woofer, and a crossover circuit for the tweeter are hooked up together inside the speaker (wired in parallel, hence the term parallel crossover is used for this type of network) and one set of terminals are present on the outside of the speaker box. In a bi-wire capable speaker, these two crossover sections are electrically separated, and a separate pair of terminals made available on the back of the speaker, one pair for the woofer, and one pair for the tweeter.

At each one of these separate pair of terminals, the LOAD seen by the amp is different: for the woofer, the majority of the LOAD is in the bass, with little loading in the treble for that separate circuit and driver. If you hooked up just one cable to the woofer terminals, then the current draw from the amp would be almost all in the bass region, with little or no current draw in the treble region. The woofer crossover is high impedance on the input to the woofer at high frequencies, and hence does not draw very much current. Compared to the traditional speaker system, with it's two crossover sections in parallel, this single cable to just the woofer section would only tend to draw current from the amp at low frequencies, while the full range crossover would draw current all across the audio band.

If we were to hook up a speaker cable to just the tweeter section on a bi-wired speaker, then this connection would draw very little current in the bass, as the tweeter crossover would be high impedance in the lows, while in the highs, where the tweeter provides output, there is some current being drawn from the amp. Again, just the tweeter section of the crossover and the tweeter only tend to draw significant current in the high frequencies, and very little in the bass.

Now, if we hook the two sections back together AT THE SPEAKER, we have essentially provided the traditional speaker/crossover hookup, and the single speaker cable will once again carry current at all frequencies, not just the bass, or not just the treble.

HOWEVER, if we run a separate speaker wire to each of the bi-wire terminal pairs at the speaker, each cable will now carry a different signal than a single cable, as the load at each pair of terminals, the woofer pair, and the tweeter pair, is different. The cable from the amp to the woofer will carry a lot of current in the bass, but hardly any in the treble region. This is a direct consequence of the way the crossover functions, and the fact that a dynamic speaker needs current to work.

Note that even if the cables were zero resistance, and zero inductance, etc., they would still carry different signals, due to the differing current draws vs. frequency.

This ties in with the diagrams at:
http://www.geocities.com/jonrisch/page9.htm

Which shows first the current draw of a single speaker cable, and then the current draw through a set of bi-wire cables connected to the same speaker, only bi-wired.

Because real world speaker cables do have some resistance, and do have some inductance, they will exhibit voltage drops bassed on this current flow. The voltage differences would be quite small for low DCR/low inductance cables, but still present nonetheless. The variations in the voltage portion of the signal may be -40 db or -50 dB or more down, but this is not as low as one might think, and since the IM distortion that would arise due to bi-wiring is not at any single frequency, but will be occuring at many different frequencies. This raises the potential audibility of the total amount of IM that occurs, and makes it more likely to intrude into the musical presentation.

Again, see the web pages I reference.

There is no doubt that differing currents are flowing in the two cables of a bi-wire set-up, and since current flow sets up a magnetic field, and magnetic fields interact with current flows, the oportunity for IM and other deleterious interactions is present.

The IM part has been measured, and I present this on my web site.

Start at :
http://www.geocities.com/jonrisch/page7.htm

Perhaps this will all make more sense to you now.

Jon Risch

Bi-wiring Attenuation of Inter-Driver IM

Fig. 1 shows a typical single wired speaker system, with a second order (12 dB/oct) crossover included. The speaker crossover components Lw, Cw, Lt and Ct are inside the speaker.

The lumped sum cable electrical parameters are shown as L1, R1, and C1.

The amplifier output impedance is shown as Ra.

Fig. 2 shows a bi-wiring connection, with the additional circuit elements it has.

Fig. 3 shows the woofer and tweeter as AC generator signal sources.

Fig. 4 shows the simplified circuit diagram for analyzing the bi-wire vs. the single wire.

Fig. 5 shows the attenuation of inter-driver IM due to bi-wiring when the second bi-wire cable is adding a second cable same as the first (a common way folks bi-wire).

Fig. 6 shows the attenuation of inter-driver IM due to bi-wiring when the single wiring has an equal resistance.
This is the way the cable naysayers insist that it be compared.

Simplifications and assumptions:
Cable capacitance is going to be negligible from a signal attenuation standpoint. This does not mean that it has no audible effect, just that it does not figure into these simple first approximation calculations.

Cable inductance is going to be negligible from a signal attenuation standpoint in the typical crossover range of a two-way speaker system. For some really high inductance cables, this will not be a good assumption, as they are significantly inductive down to 3 kHz.

Connection resistance is considered to be negligible.

So we are going to look at the simple resistive attenuation that bi-wiring provides vs. a single wiring circuit, given that the two speakers can both act as generators and attempt to send spurious signals to one another. Generator source resistance is taken to be 8 ohms, as is the effective load resistance of the other driver. Cable length is a reasonable 15 foot length for both a single wired speaker cable, and for each of the bi-wire cables.

First analysis based on the common practice of adding a second cable the same as the first one. This has the total bi-wiring resistance half that of the single wiring,

Second analysis based on maintaining equal resistance between the single wire case, and the bi-wiring case.

The table does not include the effects of the crossover components, that is, it is shown as a broadband resistive attenuation.
In practice, the loudspeaker system crossover will tend to limit the attenuation effects to around the electrical crossover frequency, for about an octave or so on either side.

Some might think that the presence of the crossover components will totally throw this resistively based analysis off. While it will change the numbers a bit, the fact that I used the nominal loudspeaker load resistance of 8 ohms, means that for real world speakers that tend to actually run lower than that, the amount of extra impedance added by crossover components that are at approx. a similar impedance as the drivers, which when added to the real world impedance of the drivers at 6-7 ohms, totals about 10-12 ohms, not too far off from the 8 ohms nominal. So for the region of interest centered around the crossover, these numbers will do for a first approximation.

This does mean that the effects of such "cross-driver signal attenuation" will be limited to a region of about 2 octaves centered on the electrical crossover frequency point.

The crossover roll off itself may be considered as part of the reduction, but this roll off is applied to

both the signal coming from the tweeter bi-wire speaker cable AND to the single speaker cable, so that any advantage this represents will be present for both cables equally.

FIG. 1
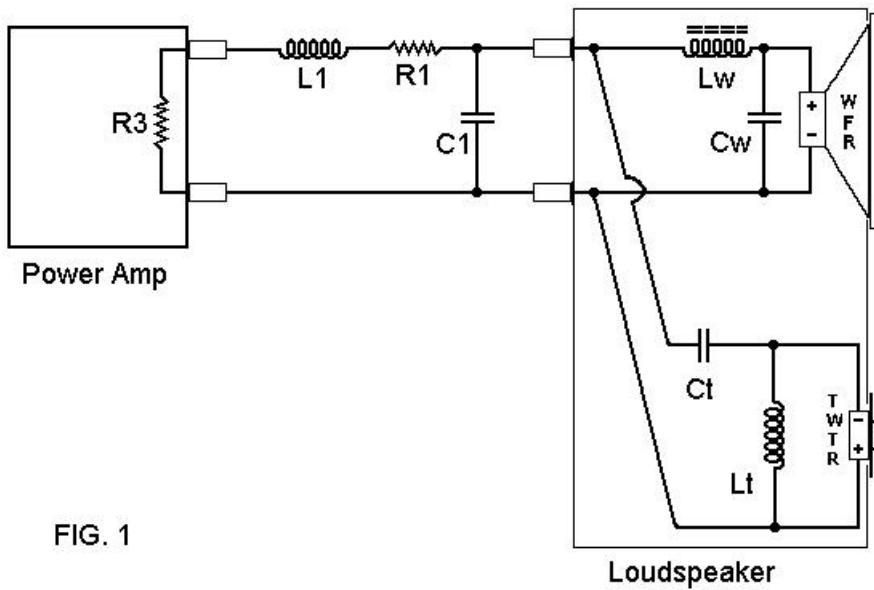
Single Wiring Circuit Diagram
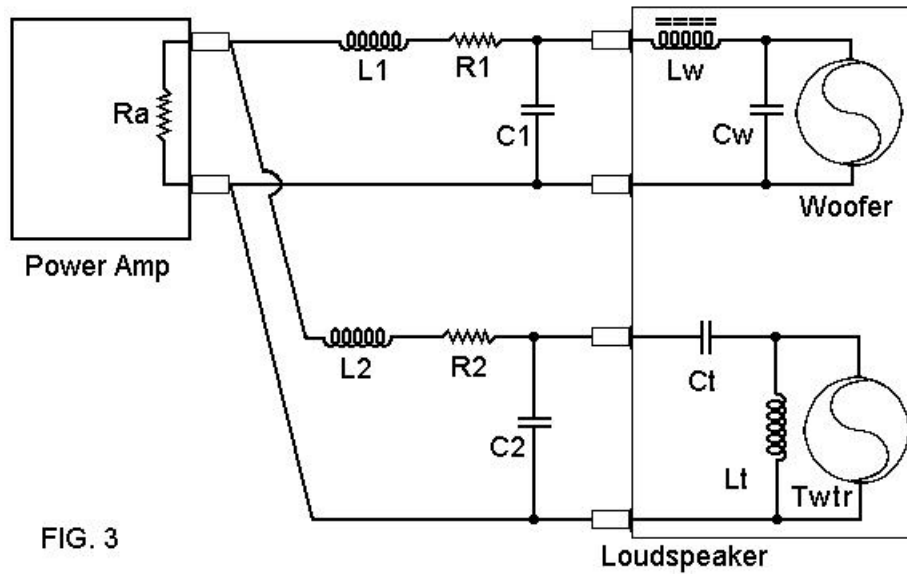

FIG. 2
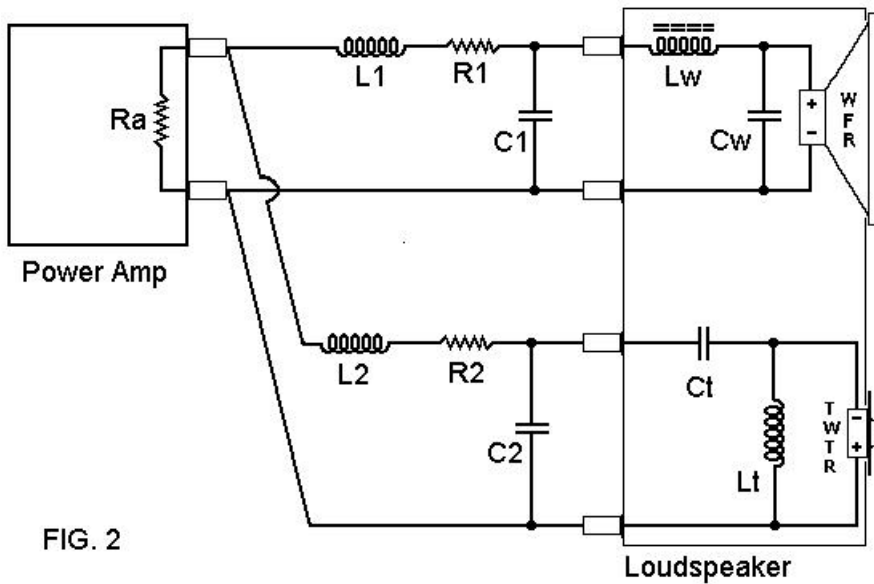
Bi-Wiring Circuit Diagram

FIG. 3

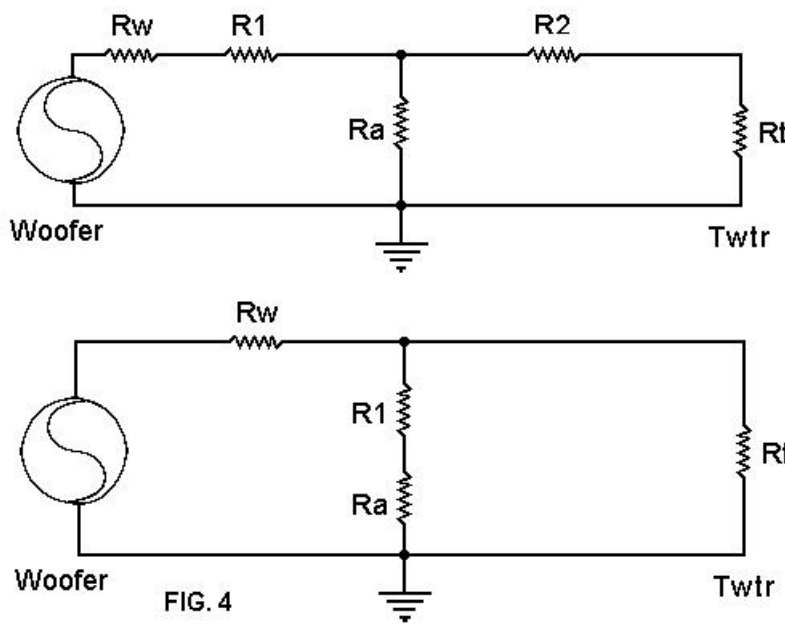**Woofer and Tweeter shown as signal generator sources**

FIG. 4

Simplified Circuit for Analysis, Bi-Wiring Circuit on top, Single Wiring Circuit on the bottom

Only the woofer into the tweeter portion is shown, the tweeter can also attempt to send a signal into the woofer.

Driver to Driver Attenuation in dB for Various Conditions

Bi-Wiring Half the Resistance of Single Wiring

Wire Ga.	Amp Output Z Ohms	DF	Bi-Wire Attenuation dB	Single Wire Attenuation dB	Difference in dB
12	0.008	1000	-60.1	-42.7	-17.4
12	0.8	10	-21.6	-21.1	-0.49
16	0.008	1000	-60.0	-35.5	-24.5
16	0.08	100	-40.4	-32.0	-8.40
16	0.8	10	-21.8	-20.5	-1.34
22	0.008	1000	-61.0	-24.6	-36.9
22	0.08	100	-41.3	-23.7	-17.6

FIG. 5

1st Set of Conditions - Adding a second (same) cable for bi-wiring

Driver to Driver Attenuation in dB for Various Conditions

Single Wiring Same Resistance as Bi-Wiring

Wire Ga.	Amp Output Z Ohms	DF	Bi-Wire Attenuation dB	Single Wire Attenuation dB	Difference in dB
12	0.008	1000	-60.1	-47.6	-12.5
12	0.8	10	-21.6	-21.4	-0.27
16	0.008	1000	-60.0	-40.9	-19.1
16	0.08	100	-40.4	-35.1	-5.27
16	0.8	10	-21.8	-21.0	-0.82
22	0.008	1000	-61.0	-30.0	-31.0
22	0.08	100	-41.3	-28.1	-13.2

FIG. 6

2nd Set of Conditions - Single wiring resistance same as Bi-wiring total resistance.

If we examine the results of this simplified first approximation analysis, we see that use of small wires for speaker cable, as sometimes practiced to keep skin effect issues to a minimum, tends to increase the attenuation of the inter-driver IM.

Very low output Z power amplifiers also tend to increase the attenuation of the inter-driver IM.

However, high output Z power amps used with large speaker cable wires having low resistance shows very little improvement over single wiring. This does not mean that all such power amps, which include most tube amps, and certain low negative feedback SS amps, will not respond at all to bi-wiring, as this is only one aspect of several with regard to bi-wiring differing from single wiring.

Source of Signal from One Driver Into Another

What are the sources of the speaker drivers feeding signal back into the crossovers? Well, there are two sources.

One source would any resonant behavior in a given driver, whatever the reason for the resonance. Woofers have a fundamental resonance that is a single peak in a closed (sealed) box, or a double peak in a vented enclosure, but they also have some cone and suspension related resonances at higher frequencies too. These often show up as small wrinkles in the impedance curve, that correlate to small bumps or a bump/dip in the speakers response, and these can be seen in the amplitude response by using a time domain waterfall response function.
Tweeters have a fundamental resonance well within the audio band, often just one octave or less below the crossover frequency. This is a prime candidate for a spurious signal as well.

The other source is the output from the OTHER speaker, that is, the woofer feeding into the tweeter, and vice versa.

Bi-wiring

Now for the relevant data regarding how much the drivers in a multi-way speaker system affect one another.

First, the details.

Data was taken on a two-way 8 ohm nominal, 8" woofer, 1" soft dome tweeter system, with a second order crossover network.
Drive level to the speakers was approx. 1W, 2.83V Curves were taken in an anechoic chamber.

The graphs have 0 dB at 0.316 Volts RMS.

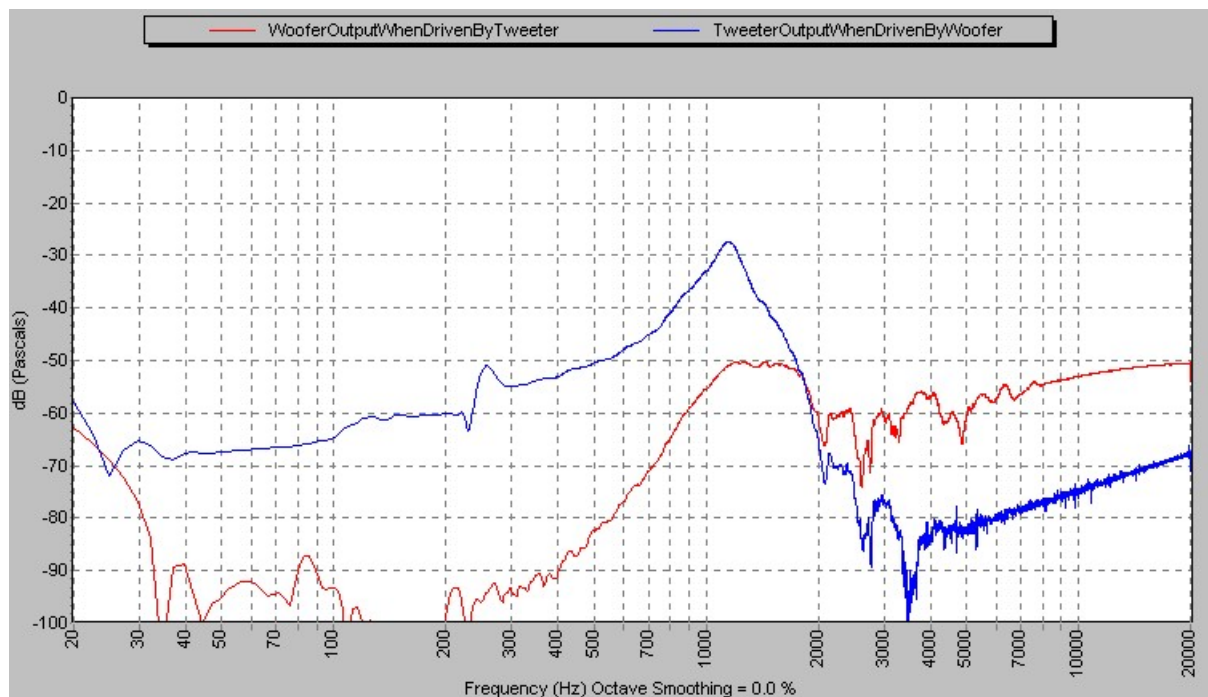In order to correlate the drive level with the output level, subtract 19 dB.

FIGURE A
Output of Woofer when acoustically driven by Tweeter in Red.
Output of Tweeter when acoustically driven by woofer in Blue.
Direct connection to driver not electrically driven (crossover bypassed).
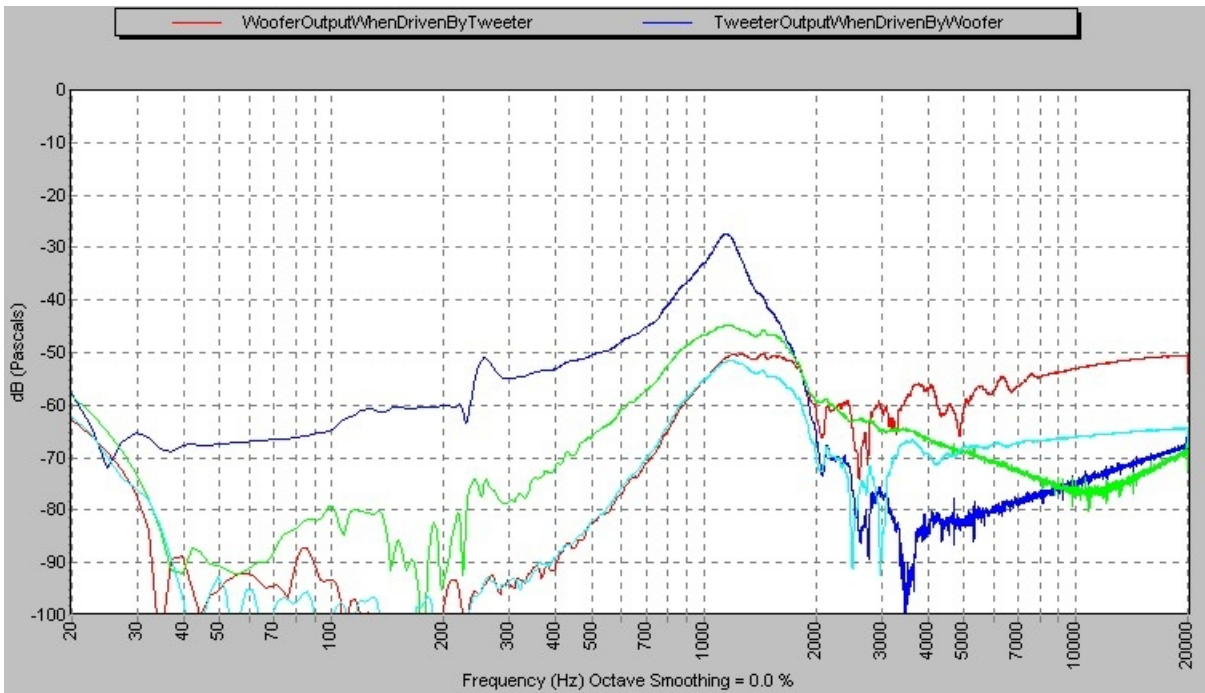
FIGURE B
Output of Woofer when acoustically driven by Tweeter, through 2nd order crossover in Light Blue.
Output of Tweeter when acoustically driven by woofer, through 2nd order crossover in Green.
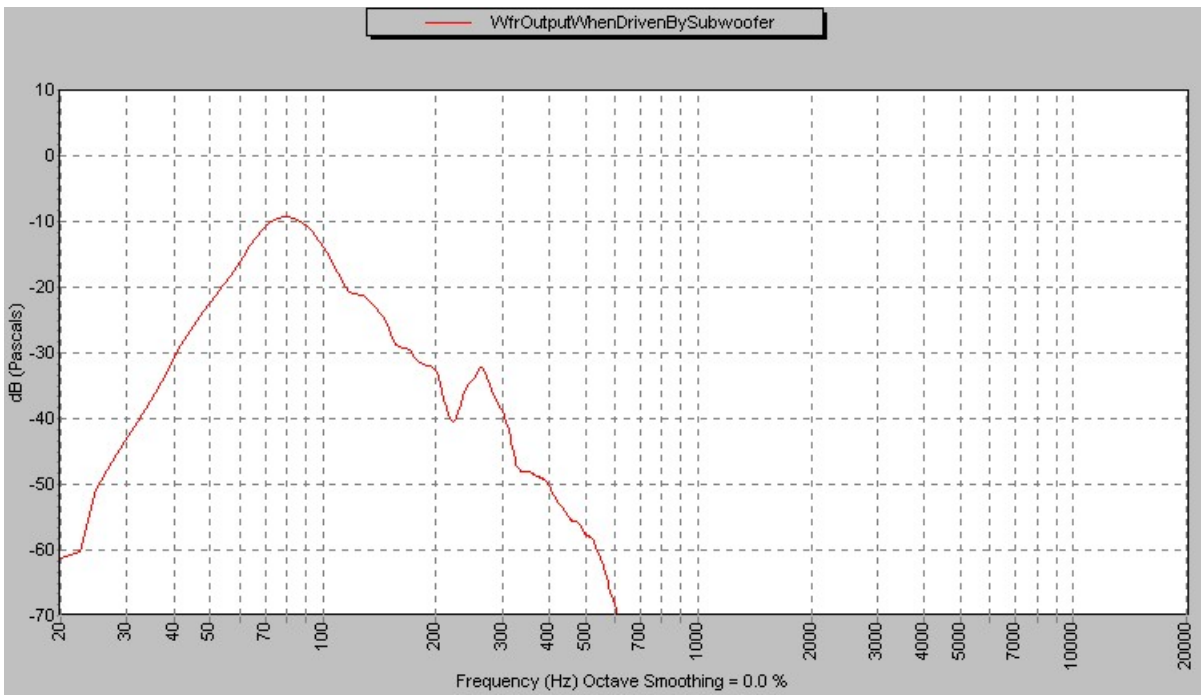Connection through crossover (crossover not bypassed).

FIGURE C
Woofer output through crossover when acoustically driven by an 18" subwoofer.
This is a Pro Sound subwoofer, with 95 dB sensitivity in the bass, so subtract an additional 6 dB
from the graph level to equate to home hi-fi subwoofer.

Analysis

Figure A indicates that the voltage coming out of the woofer when driven by the tweeter is approximately 69 dB down from the input drive level. The voltage coming out of the tweeter when driven by the woofer is approximately 47 dB down from the input drive level.

Figure B indicates that the voltage coming out of the woofer through the crossover when driven by the tweeter is approximately 70 dB down from the input drive level. The voltage coming out of the tweeter through the crossover when driven by the woofer is approximately 64 dB down from the input drive level.

These levels might seem to put the amount of inter driver IM down to levels that are below audibility, and for some that believe that distortion has to be higher than -40 or -50 dB, this would seem to be the case. These levels would then be attenuated by the figures in the driver to driver attenuation charts.

However, those familiar with my dither algorithm argument will realize that I believe that if signal aberrations are above -90 dB, they might be audible.

Figure C is the telling one, as the levels are NOT that low. Using the -19 dB to equate input level to output level, and subtracting another 6 dB for a typical hi-fi subwoofer/woofer, we can see that peak levels reach - 34 dB. This is much higher than one might have expected. This means that any system with a subwoofer will be modulating the woofer to act as a significant signal generator, and this is what could feed into the tweeter through the crossover network. Note that the curves were taken in an anechoic chamber, and so, room modes are not included, these would tend to cause peaks and dips in the overall bass levels that existed within the room, and what levels the various drivers would experience.

This got me to thinking, there is one other source of external acoustic output: THE OTHER SPEAKER/WOOFER IN A STEREO PAIR. This output would not be rolled off at higher frequencies that are still within the range of the woofer output.
If we allow for the distance from the other woofer to arrive at the levels that would occur, we would subtract another 10-12 dB for a 10-12 foot distance from the other woofer in a stereo pair. This has the peak level now at approx. -45 dB due to acoustical output from the other woofer! Due to the acoustic travel delay, this other signal would NOT be coincident with the original bass signals, and so, would constitute a definitely undesirable signal aberration. Delay would be on the order of approx. 3-4 mS.

So now we return to the tables of crossover attenuation with the various wiring schemes, and see that if we single wire, the levels of these distortions might be high enough to hear, and that by reducing them through bi-wiring, they would become less audible, or even fall below audibility.

So the action of the various speakers in a multi-way, multi-channel system WILL likely have an influence on the sound, and bi-wiring can be shown to have a potentially positive effect in some cases.

As seen in the tables Fig. 5 and 6, the amount of inter-driver attenuation will vary depending on the amplifier source impedance, as well as the effective wire gauge used for the single and bi-wiring. This explains why some folks have very positive results with bi-wiring, and some hear no changes at all, it is very much system dependent.

Further research will continue into this aspect of speaker system behavior, and I will post the results as they are obtained and verified.