Subject: Comb Filtering Misconceptions
Posted by AudioFred on Fri, 02 Oct 2009 12:56:25 GMT
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In reading all the posts on this forum over the years I get the impression that comb filtering is a potentially serious issue with line arrays whose ctc driver spacing is too great. In his white paper on line arrays Jim Griffin describes the criteria for determining appropriate driver spacing to avoid the comb filter effect. These criteria are based on the simple laws of physics as they apply to sound, and these laws cannot be violated without consequences. This I do not dispute. What I do question is some of the assumptions about what you will actually perceive when comb filtering is present, and how seriously these perceptions will interfere with the illusion of reality. How does comb filtering sound, and how bad is it?

As the article in the link below indicates, comb filtering is a reality in all two channel reproduced sound, and moving either the microphone or one's head only a fraction of an inch can result in significant peaks and nulls at the higher frequencies. Our ears can clearly hear this, yet our brain tends to "filter" it out of our perception. No listener sits perfectly still in the sweet spot while listening, but even the most obsessive compulsive audiphiles don't complain about the comb filter effect on a high frequency sound when they move their head slightly.

http://forum.ecoustics.com/bbs/messages/34579/572305.html

This is especially important as it applies to line arrays. I have listened to Roger Russel's and the Audience full-range-driver line arrays and have deliberately moved up and down while seated in the sweet spot to determine whether I could hear comb filtering while ordinary music was playing. I could not. What I did hear, in spite of the equalization that's used with these speakers, is an attenuation of the very highest frequencies, resulting in a perception of less "sparkle" in the percussion sounds. This is especially true when the speakers aren't pointed directly toward the listener. However, with these equalized full range driver line arrays I heard no more hf attenuation than I hear with most single driver point source speakers using Fostex, Lowther, CSS, and other FR drivers.

So to me the important issue isn't whether a line arrays with too much ctc driver spacing will sound bad, it's really an issue of whether one is willing to sacrifice the "sparkle" of a tweeter array in return for the other benefits of a full-range-driver line array. This may be especially relevant for geezers like me who are medicare-qualified and who can't hear above about 15Khz anyway.

Subject: Dense Interference
Posted by Wayne Parham on Fri, 02 Oct 2009 18:24:48 GMT
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Look at the concept of dense interference. When several acoustically distant sound sources (direct and/or reflected) combine, some constructively, some destructively and some in-between, what you have at any given listening position is neither a full cancellation (notch) nor full constructive summing (+6dB per doubling). Instead, you have an averaging effect. This is more like statistical energy distribution than it is like wave summing. It isn't two waves combining to make a clearly defined summed wavefront, larger or smaller depending on the phase between the two sources, rather it is more like the pattern formed by rain drops on the sea.

This is why an array sounds like it is attenuated at higher frequencies. At low frequencies, the drivers sum contructively providing 6dB more for each doubling of drivers. As frequency rises, as it approaches the range where distance is about a wavelength, clearly defined comb filter patterns begin to form. There are positions where cancellation notches form adjacent to positions where there are peaks from constructive summing. Those lobes and nulls are clearly audible as the listener passes through them. But as frequency goes higher still, the distance between lobes and nulls becomes so closely packed together you can't tell them apart from one another. They start to act more like an averaged sound field. This is dense interference, and all you really notice is a slight reduction in amplitude compared to the low frequency level, where summing is constructive.

Dense interference is what smoothes room modes when using multiple subs and it is also what causes the floor bounce notch to be mitigated by a standing line array. In a sense, what you are doing with these arrangements is avoiding the middle region where lobes and nulls are clearly distinguishable. While I think the best goal is to have constructive summing from a tight grouping of sound sources that are acoustically close, where that's not possible (indoors), dense interference is the next best thing. Since reflections make phantom sources that are acoustically distant, I see some benefit in using dense interference in some cases to smooth the average sound field, especially at low frequencies where lobes and nulls are far apart enough to be clearly distinguishable.

Subject: Re: Dense Interference
Posted by Keith Larson on Sun, 04 Oct 2009 14:46:52 GMT
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Here is an example that might be of interest. It shows how sensitive our ears are (NOT) to phase. It shows that what we do perceive is the in-air superposition of interacting sources.

Constructive and Destructive Interference, Is Phase Audible or In-Audible?
http://www.woofertester.com/phase_audibility.htm

Best regards,
Keith Larson

Subject: Audibility of Phase
Posted by Wayne Parham on Sun, 04 Oct 2009 16:57:13 GMT
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This has been a sort of hot-button topic of mine, over the years. There are a few manufacturers

that have made "time alignment" a marketing catch phrase. It always sort of irks me. I prefer to discuss the position of lobes and nulls, because I think it is more accurate to talk in these terms for several reasons.
Posts about Phase AudibilityI suppose it's really a semantics thing. But what I have noticed is that some manufacturers artificially distinguish themselves from others using a marketing ploy, saying their product is "time aligned". The concept used by most of these guys is the same thing suggested by Altec in the 1960's and 1970's. No better and no worse. So when you see someone champion the virtues of time alignment, look closely at what they're trying to say. Ask yourself if they're really setting the position of the forward lobe or if they're just trying to impress their audience with gee-whiz words.
Altec Application Note 9, "Polarity and Phase"The problem is each sound source has a physical position, so there is a specific path length from each source to the listener. This path length may change as the listener moves, and it may be different between two sources in a loudspeaker, either from two drivers in a multi-way loudspeaker or even at different points across the surface of the radiator of a single driver. There is also the matter that electrical, mechanical and acoustic reactances causes phase that changes with respect to frequency. These two things - delay from physical offset and delay from phase shift - are not the same thing. One changes with frequency and one does not. So using one to counteract another works only over a very narrow band of frequencies. It is a worthwhile approach, but the point of all this is it means "time alignment" is a relative thing (relative to position) and it probably makes more sense to talk about the position of the lobes and nulls.

Keith's example in the previous post does a great job at illustrating the point that our ears are much more sensitive to the changes in amplitude response as a result of combining signals with phase shift than they are to the phase shift itself. Another example of how our ears interpret time delays of this sort is to listen to a sawtooth wave with the peak on the left verses an identical one with the peak on the right.

These two waveforms have exactly the same frequency harmonic content but have different phase. The difference between the two is reverse sawtooth waves have even harmonics phase shifted. Like the headphone test in the previous post, listening to an individual waveform in isolation prevents the listener from obtaining frequency cues that expose a difference in phase. No summing is taking place. If a person could detect phase shifts even without a corresponding change in amplitude response from summing, they should be able tell the difference between a sawtooth wave and its reverse. But they cannot.

Cancellations of certain frequencies, beat frequencies and things that make response aberrations are pretty easy to detect, but absolute phase isn't. So you're going to have a very hard time finding people that can detect a sawtooth from a reverse sawtooth in a blind test. Try it and see.

Subject: Re: Audibility of Phase
Posted by Keith Larson on Wed, 07 Oct 2009 21:32:25 GMT
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The way I understand how the ear works is that the cells that are packed in the cochlea tube are tickled linearly along the tube path depending on the excitation signal wavelength. This is somewhat like a spectrum analyzer, except it is amplitude, not the phase that is tickling the neurons. Reversing the polarity of both your speakers should make no difference in the sound.

On the other hand, if a driver is pushed hard enough it will distort. In particular, as the voice coil comes out of the gap BL, Q, inductance etc. are all changing. Using a two tone sine wave test will result in a weird effect that sounds like a doppler effect that is independent of driver polarity. On the other hand, a highly polarized test signal like a sawtooth results in a different sound (ie timber or tonality) as the driver is sucked in or out depending on connection polarity. I see this regularly when testing drivers at higher power. If you think about it, a kick drum will also have some polarity to it. So, it should be no surprise that someone could pick out an A-B difference.

Best regards,
Keith Larson

Subject: Re: Audibility of Phase
Posted by Wayne Parham on Thu, 08 Oct 2009 02:30:49 GMT
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You're right about the condition that exists at high drive levels. You and I have discussed this before. At extreme levels, where a driver becomes non-linear, some (most) act differently in one direction than the other. I've seen that in woofers in particular, and I imagine it happens in most kinds of drivers.

But this is an emergent behavior. It isn't that anyone can hear the phase, what they're hearing is a non-linearity at extreme levels that manifests itself being audibly different depending on polarity. The distortion causes internal summing, modifying the amplitude response depending on phase. What's audible is the nonlinearity, not the phase.

I agree with you on this condition, if a driver were pushed well beyond its linear range, a person may very well be able to hear a difference between a sawtooth and a reverse sawtooth. They might be able to tell the difference in a signal presented to it in reverse polarity. But again, what is being heard isn't a difference in phase, it's a difference in spectrum, a result of a non-linearity in the system.

Subject: Re: Audibility of Phase
Posted by Keith Larson on Thu, 08 Oct 2009 15:28:12 GMT
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We are definitely on the same path and these are good words for refining the point. I guess another way to state this would be that speaker POLARITY effects both phase AND non-linear distortion. Since our ears are phase agnostic, this leaves the distortion.

BTW, one of the things about distortion analysis is that it does not clearly indicate a polarity problem. And, it wont clearly show stuff like crossover distortion either. Bandpass filters, and simple FFT spectral analysis are especially prone to this because they are primarily interested in the sine wave energy (there are some more obscure FFT techniques that can be used). An oscilloscope view would be the better tool, but the distortion needs to be on the order of 10% to be noticed. I addressed this with one of the XY views in the WTPro by locking on to and subtracting the fundamental from the raw data, leaving just the distortion.

Subject: Audibility of Non-Linearity
Posted by Wayne Parham on Fri, 09 Oct 2009 01:01:45 GMT
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I'd like to know more about that. Please elaborate if you have time.

For right now, I have three ways of looking at non-linearity:

1. With an oscilloscope, look for clipping. This is easy to do if the signal is electric.
2. Look for harmonics using a stepped sine. Lock onto whatever multiple of the fundamental I want to see, 2nd, 3rd, etc.
3. Use two sines and watch for sum and difference tones.

Each of those tests expose something about system non-linearity, so I can learn something from each one.

I always welcome new information, to explore other things that might prove useful or interesting, so please explain your thoughts when you have time.

(We may want to move or split this thread into the measurement forum, because I think it's headed more and more that direction.)

Subject: Re: Comb Filtering Misconceptions
Posted by Keith Larson on Sun, 11 Oct 2009 15:09:18 GMT
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Getting back on track for comb filtering, I could point out a number of highly successful commercial systems where interference patterns were intentionally created for a sense of realism.

Back side and cross firing drivers producing L+R, L-R etc. were quite popular at one time. The advertisements would even tout the effects of comb filtering as a positive effect. Nowadays there are electrostats, dipoles, bi-poles, and line arrays. There is even a realization that speaker placement has an effect on comb filtering.

As a half fact half subjective observation, I would point out that comb filtering in the 300-3k intelligibility band is probably the most noticeable. On the other hand, we don't often listen to high frequency sine waves. In fact, most high frequency content is found in transients like the crash of a cymbal, the ting of a bell, or noise like 'S' sounds.

What is interesting is that when you examine these kinds of signals in the frequency domain they are highly complex in frequency, time and amplitude, but the duration is usually short. This is where Fourier analysis over a wider (often vastly) time window does not produce the right results. Technically, the FT is produces an observation of the average content over a finite length window of time. If the signal is a transient it splatters all over the spectrum. The solution is to use a sliding or overlapping window and watch the response take shape like a movie.

For me at least, its difficult if not imposable to directly hear comb filtering in a transient signal, especially when it is above the intelligibility band, and maybe this is the key. In the end however, this is technically a coloration, so its a matter of taste. Some will like it, others will not.

Subject: Re: Comb Filtering Misconceptions
Posted by Wayne Parham on Sun, 11 Oct 2009 16:58:23 GMT
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I think I see what you're saying. In way, you are saying it might make sense to look at the sound field averaged over time. Sort of like dense interference uses multiple acustically distant sources spread out in space, your concept is to look at them spread out over time as well. Then another thing you suggest is that comb filtering may sound interesting, in some cases. I do know that Moog (inventor of the electronic synthesizer) suggested this exact same thing.

Subject: Re: Comb Filtering Misconceptions
Posted by Keith Larson on Mon, 12 Oct 2009 03:29:06 GMT
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Yes, kind of. I was heading down the path that high frequency transients don't really have a solid well defined spectrum.

Suppose a transient signal that occurs left of center. The left ear gets a clear shot, but not the right. As the sound wraps around the head, combing and shading effects that ripple with time are coming into the right ear. There are probably some fast triggering A before B neurons in there, but Ill also bet spectral shimmering has an effect as well. Who know, this might lead to another interesting headphone -vs- speaker test.