Subject: Phase Delay and Group Delay
Posted by Farb Sklarb on Thu, 08 Aug 2002 19:40:35 GMT
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This is a follow-up to the earlier discussion on crossover correction by time alignment. I'm not sure a purely theoretical posting is of much interest to most readers of the forum, but for some reason I'm in the mood for stirring things up, so what the heck.In the previous discussion, it was mentioned that the phase shift through the crossover introduces a delay. The value of this delay, which is called phase delay is proportional to the negative of phase shift divided by frequency. The phase shift in this case can be observed as an offset in the voltage peaks between the input and output of the crossover network.The effect of driver reactance on phase shift was mentioned, but this is really a secondary consideration. In speaker design, the goal is to achieve a target system response which takes into account the electrical interactions between the crossover and the driver, as well as the acoustical response of the driver itself. So we can ignore all that complexity and approach this phase delay question purely in terms of the relationship between the acoustic signal coming out of the speaker driver and the electrical signal at the input to the crossover.In the case of the LF driver, the ideal first-order crossover has a phase that starts at zero degrees at DC, is relatively flat up to a couple octaves below the crossover frequency, declines smoothly to minus ninety degrees, and is relatively flat again out to infinity. This is evident if the response is plotted logarithmically, which is usually the case. The graphs in the earlier discussion were plotted against linear frequency axes which hides some of the nice symmetry of the phase response. Due to the minus sign in the equation, the phase delay associated with this target crossover function is everywhere positive and there is apparently no philosophical conundrum since the output of the crossover clearly arrives after the input.In the case of the HF driver, the phase starts at ninety degrees and varies to zero, being positive at every frequency. This means the phase delay is everywhere negative. If phase delay is physically meaningful in the conventional sense of cause and effect, then this leads to the absurdity that the output of the speaker arrives before the electrical signal that produces it. Again, this has absolutely nothing to do with driver reactance or the capacitor in the crossover network. It's a fundamental property of the electrical-to-acoustic transfer function of the tweeter channel of the loudspeaker.We could replace the loudspeaker with a communications channel having the same phase and frequency response, and the result would seem to be a device for sending messages backwards in time. This violates fundamental principals of communications theory and tells us that phase delay is not the same as "delay" as we commonly think of it.What this tells us is that there is no causal relationship between, say, a peak of the AC signal waveform at the input to the crossover and a peak of the acoustical signal waveform at the output of the speaker. If there were, we could use this relationship to encode messages going into the past through phase-lead networks. For continuous, unvarying sinusoidal test signals we cannot send any information over this channel. And we also can't measure the phase shift because we have no way of knowing how many cycles have "slipped" between the input and output.In fact, the only way to measure the phase shift, which is also the only way to send a message over the channel, is to modulate the signal. For instance, we can abruptly shut the input off and see how long it takes for the output to respond. A more methodical way to make this test would be to mix the input signal with another sinewave having a very slightly different frequency. The result of adding these two signals would be a "beat note", which is a form of amplitude modulation. With a bit of work, we can measure the time it takes for the modulation to make it through the system. We will find it is not equal to the phase delay, but to the negative slope (i.e., change with respect to frequency) of the phase-versus-frequency curve. This is known as group delay, and it must be positive for all causal (i.e., realizable) analog filters.  Group delay is what determines the observable delay of systems like filters.  It follows that the phase shift must always decrease (become more negative) as a function of frequency for all crossover filter functions, else we are violating causality by sending music backwards in time.Music is reversible, time is not.  Go back... Go back...A delay line is, in fact, a type of filter, having uniform frequency response and linearly decreasing phase. Because the slope of phase for a delay line is constant, the group delay is constant, and so signals passing through it are delayed equally regardless of frequency.  Physically offsetting a driver on the baffle is equivalent to adding a delay filter to the crossover function.  It can't "correct" the phase at all frequencies, but it can be used to adjust the group delay over a range of frequencies.  I can think of several cases where this could be a useful trick for crossover design.Apologies in advance for any errors in this message.  I have an interest, but lack expertise, in the subject matter.  Corrections are invited.Have fun.

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Subject: Re: Phase Delay and Group Delay
Posted by Wayne Parham on Thu, 08 Aug 2002 20:27:16 GMT
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What we are dealing with here is a reactive circuit.  And in such a circuit, you have three components - resistance, inductance and capacitance.  Current flows through resistance at the same rate that voltage is applied across it.  But for inductance and capacitance, the rate of current change is different than the rate of voltage change.  For a capacitor, current passes through it very rapidly, but voltage rises across it slowly.  Inductors are exactly the opposite, with voltage rising across them rapidly and current flowing through them later.Since power is the product of voltage and current, this phenomenon makes a difference in how power is transferred through the circuit.  It is also important to realize that the reference point is resistance, and that reactive circuits are phased in reference to that.  In other words, one can see capacitance as leading resistance by 90 degrees and inductance lagging it by 90 degrees, but it would be just as accurate to say that resistance lags capacitance by 90 degrees and inductance lags it by 180 degrees.The most important item then is the phase angle of the power delivered to a speaker motor.  This is described on most charts relative to what the phase of the input signal when applied to a resistive load, which is fixed in phase and does not change.  In relation to this, a reactive circuit will generate a shifting phase that moves with respect to frequency.

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Subject: Re: Phase Delay and Group Delay
Posted by bmar on Fri, 09 Aug 2002 01:52:31 GMT
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how do the different order crossovers affect the phase shift?

Subject: Re: Phase Delay and Group Delay
Posted by Wayne Parham on Fri, 09 Aug 2002 02:27:26 GMT
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Higher order crossovers shift phase more than low order crossovers do. But then again, attenuation occurs much more rapidly, so by the time phase difference between adjacent subsystems is climbing much beyond low order levels, one of the two subsystems in overlap is sufficiently attenuated so as to not cause incoherent summing and ultimately destructive interference. The overlap band is reduced, and the amplitude of out-of-band signals is also reduced.There are a couple of things that contribute to destructive interference: One is physical distance between sources and the second thing is phase from the electrical filters. This is discussed in a alittle more detail in the post called "Biamping-xovers-12, 18, or 24db/oct".

Subject: Re: Phase Delay and Group Delay
Posted by hancock on Fri, 09 Aug 2002 09:52:21 GMT
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As I understand it, you are comfortable with a positive 90 degree phase shift as not breaking the time space continuum, but a negative 90 degree phase shift is more of a conceptual problem. How about trying this--apply a positive ninety degree phase shift to a signal and then reverse the polarity of the signal. Now you have a negative 90 degree phase shift, but you haven't broken any laws of physics.Hope that helps, John

Subject: Re: Phase Delay and Group Delay
Posted by Farb Sklarb on Fri, 09 Aug 2002 12:49:54 GMT
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Thanks for the note. Actually, I believe that negative phase shifts are associated with positive time delays, and positive phase shifts are associated with negative time delays. So I have a simple rhetorical question. If a filter introduces a positive phase shift between input and output, knowing just the frequency and the phase shift, do we have enough information to say how long it takes that signal to pass through the filter?:-}

Subject: Re: Phase Delay and Group Delay
Posted by hancock on Fri, 09 Aug 2002 15:45:14 GMT
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You are absolutely right, I always get the sign of phase shifts mixed up.Regarding your second question, the simple answer is no, the pedantic answer is in a very hypothetical circumstance, it might be possible. If the filter is an analog filter, then the answer is easy: in theory it takes forever for the signal to work its way through the filter unless it's a simple gain.For digital filters it's a little more difficult. There are two basic types of digital filters, Infinite Impulse Response (IIR) and Finite Impulse Response (FIR) filters. As the name suggests, it takes an infinite amount of time for a signal to work it's way through an IIR filter. The time it takes for a signal to work its way through an FIR filter is equal to the order of the filter.In engineer speak, IIR filters have at least one pole and can have many zeros. FIR filters have only zeros. Your question then amounts to "is knowing phase without knowing magnitude enough to a) know whether or not the filter has any poles and b) know the number of zeros in the filter?"If all you have is the phase from a Discrete Fourier Transform (DFT) calculated from a finite impulse response measurement, then the answer is no you can't tell the number of poles and zeros simply because it would take an infinitely long impulse response to do so and you don't have an infinite amount of time to make take that sample.On the other hand, if you know the phase from a Fourier transform of the filter transfer function, it may actually be possible to at least tell if it is an IIR filter or not. I don't have an immediate answer to that question. This is, of course, a hypothetical question, because if you know the transfer function of the filter, then you know how many poles and zeros it has.I hope that helps more than it confuses, but am pretty sure it won't...John

Subject: Re: Phase Delay and Group Delay
Posted by Wayne Parham on Fri, 09 Aug 2002 19:10:44 GMT
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The behavior of reactive circuits is pretty simple stuff really.  Power developed across a speaker motor is phased - moved in time - by the reactive behavior of the crossover circuit, and any other filters in the system.  The problem is that phase moves with respect to frequency, and therefore, so does the amount of delay.  The delay wouldn't be so bad if it were uniform, but it isn't.  So interactions between components that have been phased differently can cause cancellation of selective frequencies in some cases, which causes response anomalies.Adjacent second-order networks are an excellent example of this phenomenon, and their behaviour is easy to understand.  When a woofer uses a second-order Butterworth low-pass filter, it's apparent distance is shifted further from the listener (than it's actual position, or "where it is" at very low frequencies) by exactly 90 degrees at the -3dB crossover point.  The distance is equivalent to 1/4 wavelength of the crossover frequency, which is a specific amount.The tweeter, using a second-order Butterworth high-pass filter is shifted nearer to the listener - again, referenced to it's actual position, or "where it is" at very high frequencies - by exactly 90 degrees at the -3dB crossover point.  The distance here too is equivalent to 1/4 wavelength of the crossover frequency.This makes the total phase difference be 180 degrees, which is 1/2 wavelength.  It is also a "bad mode" where destructive interference is maximized.  So in this case, the difference in the phase of the power applied to the motors is presented at a different time, it has moved by exactly 1/2 wavelength and the diaphragms then move delayed from one another in a fashion that cancels each other out.  That's why second-order networks are usually "cross-connected" by reversing the connections to one of the motors.  We have a time domain issue that has manifested itself in the frequency domain.  So by cross-connecting, we have reduced the effects in the frequency domain, but we have not reduced them in the time domain.  We have only changed the conditions and found a good working solution to the problem that is causing a frequency anomaly.So the fact that there is a quantifiable time offset created by the crossover is the reason why baffle offsets are sometimes incorporated.  And there is some merit in doing so - if the apparent distance between the listener and adjacent sound sources is equal at the crossover frequency, then the idea is that it will move a little one direction above the crossover frequency and a little in the other direction below the crossover frequency, and that the position chosen is then a good compromise.But another fact is that by moving the diaphragms forward or backward, we now create a situation where specific phasing issues happen at different frequencies that where they would if the diaphragms were exactly in physical alignment.  Since the apparent distance moves with frequency, if we choose one frequency to focus on - say the crossover frequency - then we can align diaphragm phase at this frequency.  But as frequency changes, the diaphragms move out of phase with one another.  We could have chosen a slightly different offset, which brought them into alignment at a slightly different frequency than the crossover frequency instead.  Then the crossover frequency wouldn't be in alignment anymore.  And then there is also the troublesome issue that as we move at angles, we change the distances between the listener and the adjacent diaphragms. So we can set alignment for specific frequencies in specific positions, but we can't align the system overall.In my way of thinking, this makes an answer very clear:  Shoot for controlled directivity and make the target area be the region for which the system can

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Subject: Copy paste print & save
Posted by Art J. on Sat, 10 Aug 2002 16:55:19 GMT
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Hi Wayne, its been a while.We talked about this last year.You summed it up very well, I like it.C U , Art