Subject: Time alignmet vrs reality
Posted by bmar on Mon, 08 Jul 2002 22:46:13 GMT
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If an orchestra is spread about a stage some 75 feet wide. Are the instruments on one side of the stage in sonic time alignment with the instruments on the other side? I would imagine that some thought has gone into the seating arrangement so that a Piccolo is not right next to a Bass. If a person was to be sitting in the front row of a theater listening to a orchestra. Are the people in the balcony hearing an equal alignment. Both are considered good seats. If the concert was good, everyone leaves happy wishing they could have sound like that in their living rooms.I'm just curious to how time aligned the actual source is that is recorded. Then, we scrutinize time alignment to the enth degree, only to reproduce what? That would be my real question. How time aligned are the sounds that are recorded in the first place.Obviously placing a woofer 10 feet from a tweeter is improper, and a bass horn that is 3 meters long would need a touch of delay. much beyond that where is a good place to draw the line.just some thoughts to ponder (not fight about)Bill

Subject: Re: Time alignment vs reality
Posted by mikebake on Mon, 08 Jul 2002 23:14:43 GMT
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What you hear at a given point in the hall is the original sound; time alignment in the speaker simply needs to replicate what the mic heard (within the usual constraints) at it's locatoin without adding to it. I don't see a time alignment scenario in sound production, as opposed to reproduction. If one sound arrives slightly later than another in the live sound, it's normal (unless someone is off the beat.......)My idea of coming closer to ideal reproduction was not that you use two speakers or a surround system to replicate sitting in the hall, but rather that you have an ideally sized driver for each instrument in the orchestra, all in their proper original spatial relationship, and listened to in a similar hall. That would be cool, but would require multi multi track sources, etc. It might be a bit expensive and low on WAF too..........

Subject: Re: Time alignment vs reality
Posted by Paul C. on Tue, 09 Jul 2002 00:50:49 GMT
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In a concert hall,listening to live instruments, each individual instrument reaches your ears "time aligned".  Its fundamental, middle, and upper harmonics arrive to your intact.  But a speaker system breaks this into parts, ALL of the instruments suffer from the crossover effects. Fortunately, your ear is used to hearing room resonances and dips, and the brain sets up a sort of reverse filter to compensate, as much as possibleNow, the individual instruments DO come to you with slightly different times... they coordinate, or try to, with the conductor, yet their distances, to any one listener, are different.Where this can really get sticky... a drum and bugle corp or band on the football field at a ball game. As they get spread out, the tendency is for the musicians to try to listen to each other, and as they do, the delay due to speed of sound can really get things screwed up. What they must do is listen to the drum section, positioned in the middle. The drummers WATCH the conductor (drum major), and play with him/her. The drum major must have a good sense of tempo and totally ignore the sound of the band, and just conduct. Otherwise if the drum major synconizes with the band, the whole thing will get slower and slower, and bog down. I know, having been both a player and conductor in these bands, and on stage with concert bands and orchestras.Another effect you may not be aware of... close micing vs distance micing. The low frequencies travel farther, easier, than high frequencies. High frequencies in the tone, known as "edge" among musicians, gives projection to the tone. Yet, too much gives a coarse, grating tone.The overtone spectrum of an instrument changes as the volume level the musician plays is changed. It is not the same as just turning the volume knob on the amp higher or lower.When a musician plays in a large theater/concert hall, if he knows what he is doing, he plays with a bright, edgy tone. A fine symphony cellist, for example, sounds like a buzz saw up close. He MUST play with this type of tone, or he could not be heard past the third row. BUT, out in the audience, the tone will be sweet and mellow, rich sounding. The "edge" disappears with distance, leaving the mellower fundamental and middle overtones. Musicians speak of "theater tone", "studio tone", or distainfully, "parlor tone". Students and many teachers do not understand this effect. A student with a very pretty, sweet, dark tone in the practice cubical, or the teacher's studio during a lesson, cannot be heard on stage for his solos or exposed passages.It is a characteristic of musical instruments that when they play very softly, only their fundamental and first few overtones are present. As they play louder, more overtones are generated, and at higher levels. A sax, which has both even and odd overtones, playing very, very softly, will only produce the fundamental and only 2 or 3 overtones at pp (pianissimo, very soft) levels. At mp (mezzo piano, or medium soft) there will be about 7 overtones, yet the fundamental and first two overtones are no louder than at pp. At f (forte, loud), there will be maybe 14 overtones, the middle overtones stronger, yet the fundamental and first two overtones are no stronger than when played pp or mp. At ff, (fortissimo, very loud), there may be as many as 28 overtones generated, but still the fundamental and first two overtones are no louder than at previous volume levels.This is typical of most all musical instruments, when playing louder, it is in the number and intensity of overtones that make it sound louder. (source: The Saxophone Is My Voice, Ernest Ferron: similar observations by Dr. Arthur Benade in his many papers and books)So, there are many effects working in live music vs close mic'ed recorded music.

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Subject: Re: Time alignment vs reality
Posted by mikebake on Tue, 09 Jul 2002 00:56:49 GMT
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Ahh, Paul knows whereof he speaks. The theater, studio , parlor tone is a reality that I have heard musicians speak of (good ones, anyway). Thanks for the further enlightenment. Your statements are interesting and factual.

Subject: Re: Time alignment vs reality
Posted by Wayne Parham on Tue, 09 Jul 2002 01:35:16 GMT
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Yes, Paul does know the issues better than most.  He's a contributing author for Sax on the Web and is very familiar wih the nuances of sound both from a musician's and from an engineer's standpoint.

Previous discussions about "phase" and "time alignment"

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Subject: Re: Time alignment vs reality
Posted by Paul C. on Tue, 09 Jul 2002 13:05:00 GMT
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When I record live in a concert hall, I like to position the mics, JUST TWO MICS, on a double mic holder about 8" apart, and angled out about 20° each from straight ahead.  I put them back about the third or 4th row, or right in front of the orchestra pit for a band or orchestra on stage.  I jack them up about 10' or so.  This gives excellent stereo imaging upon reproduction.If the mics are spread out, the effect is rather unrealistic at home, sounding unnaturally spread.  We hear direction not by volume difference, but by phase difference, between the left and right ear.Due to the distance between our ears, and the relative length of very low frequency wavelengths, we get pretty much the same phase in each ear.  That is, both ears hear the peaks pretty much simultaneously, as well as the valleys, of low frequencies.  But for shorter wavelengths,  there can be quite a difference.Where the wave lengths are around twice the distance between the ears, the phase can be 180° out from left ear to right ear... the left hearing a peak, the right a valley.  And other frequencies, other phase differences.And that is for a single sine wave.But real sounds have all sorts of frequencies, and an orchestra, lots of instruments, lots of overtones, lots of notes.  The brain is quite a complex processor to be able to handle all of that, yet it does so quite easily. Amazing!Time alignment is most critical on transients, that is, short, percussive sounds.  When a drumhead, or other object is struck, all the frequencies produced are peaked at the time of impact, and decay from there.  If these peaks do not get to your ears all at one time, the sound is smeared.  This is why the fewer the crossover points, the better.  And not only that, position that crossover point down in a region where the wave lengths are longer, and easier to align.  I do not want time alignment , phase shifting, going on up around 8Khz-10khz.  Or even at 5 khz.  That will smear the sound.As Wayne does, a two way, with crossover in the 800hz-1.6khz range, this leaves percussive sounds pretty much intact, and the upper partials are easier to align with the lower, longer wave lengths.  So, you do not have things arriving over several cycles to the ear, but within one cycle.This is what time alignment is all about.  Good transient response.  Listen to percussion instruments... if that sounds convincing, the speaker is pretty good in the time alignment area.With sustained tones, it means nothing, but where it causes a notch or peak at crossover... a frequency anomaly.Now, the ear (really, the brain) will tolerate fixed resonances, or holes. It hears this as the ambient room sound. After a few minutes of listening to music in a particular room, the brain sets up a sort of reverse filter, and pretty much ignores this effect. Now, if it is really bad, like poorly designed speakers with a one note bass thump, yes, you notice that. But the narrower and lower those resonances, the easier the brain ignores it.This is sort of like wearing tinted glasses, after a while, you no longer see color shifts.Well, like Forest, that is about all I have to say on that subject.

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Subject: exactly, the subtle benefit is mainly
Posted by Sam P. on Tue, 09 Jul 2002 14:10:56 GMT
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for transient, percussive sounds like the tinkle of chimes and cymbals. When my Altecs are "time AND phase aligned" with the HF positioned directly above the LF, small background musicallity seems to be clearer and more distinct. Hard to describe, easy to A-B demo. When the HF is positioned 1/2 wave behind the LF(at 600Hz., about 11.25 inches physically), both drivers will again "sum" correctly at xover with both wired positive. But that less than a foot difference in physical positioning somehow is changing the character of the transient reproduction slightly. This is phase aligned, but not time aligned. I am not claiming the subjective differences are purely due to the time delay. Just that the effect is audible, under some circumstances, i.e. reproduction of "chimes, etc.". This might be something that only is audible when phase at xover is closely aligned? In the classic Altec AN-9 method of alignment, away from the xover filters phase shifts, the rest of the signal content would be passed "unshifted" and well TIMED(?), hence might "line up" all the higher freq. harmonics associated with that drum stick striking the rim better? Like a 100kHz band marker generator(square wave?) will put a "reference" signal EXACTLY at every 100kHz. harmonic on your radio dial, right. So how "nice" would it be, to take the upper half the band, and "shift" it's band markers higher or lower in freq ON THE DIAL, as would happen to upper harmonics when drivers physically have a difference in distance from the listener, and are called upon to reproduce a "series of harmonics". Hmmmnn. Very inelegant:) Even without the xovers phase shifts kinking things! One suggested album cut for "listening" to this effect is Santana Abraxus track one with the chimes. On another occaision, with a certain recording, I would notice a distinct "repositioning"(approx. 18 inches) of certain instruments like drum clicks in the soundstage when changing from the "time/phase" aligned horn position to the "phase" aligned one where the HF is 1/2 wave behind the LF at xover. Don't "shoot" the messenger, I'm just reporting OBSERVATIONS I have no way to explain definitively. But the "flying horns" needed to get time/phase alignment with my direct radiator LF is a pain(literally), the front flange of the 511 needs to be around 15.75 inches IN FRONT of the leading edge of the 4508 LF enclosures, mechanically works fine with "Sam's Time Alignment Trolley"(pat.pending.), but sux when you walk into them in the dark the first time:( So I usually listen with the HF physically 1/2 wave behind the LF, 511 front flange about 4.5 inches out front, with the 600Hz. xovers in use. Phase aligned, not time aligned. HF and LF both wired positive, FWIW. And "looks more normal" than the flying position that "is best in theory". REALITY MEETS SHE WHO MUST BE OBEYED :) Sam

Subject: Re: Time alignment vs reality
Posted by bmar on Tue, 09 Jul 2002 14:37:04 GMT
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Paul,This is exactly the kind of information I was looking for. Thanks so much for taking the time to post your thoughts on this. What you say makes sense, and for a person with out the higher level math background needed to conjugate all the formulae, this makes sense. I can see the possibility for all the wave lengths to converge as they near the listener. So just to continue the conversation, I have a hard time believing that everything is in place with such a wide range of instruments. I chose an orchestra to talk about since it has the largest amount of diversity. Much like you mentioned that people seated in different area will hear different things, or the marching band spread out across the field. What if we look at a grand piano with its lid open. I wouldn't say its a picture perfect horn, but surely its adding to the projection of the sound. Like the single driver speaker that Wayne pointed out as being impossible to be in phase with itself. Would this apply to a piano which covers 7 ? octaves. Let alone a pipe organ blowing down to 16hz at low c. I would like to see some sound and wave length measurements taking at the same place your mic setup is at the time of recording. Then, play the recorded media and see what it takes to duplicate the same measurements. Is the 2000 interconnect, the 1000 speaker wire, 40k mono block! Perhaps this kind of measuring and comparing has been done already. I would have thought it would be high on the list of an advertising champagne to offer such results other than, "is it live, or is it Memorex" ! The list goes on in what people have spent and will spend to achieve this perfect colorless sound that 100% duplicates the sounds that were recorded. It strikes me funny that a person would need a audio system worth far more that a lot of the recording gear used at the time of recording. I'm not saying any of this is bad. I'm just wondering how close to "real" is even possible to obtain. In fact I love the challenge of combining all different types of audio gear to achieve different effects, so to speak. It's great fun/business/hobby.... how ever you may be into it.And, its good discussion too. Thanks,Bill

Subject: Re: exactly, the subtle benefit is mainly
Posted by bmar on Tue, 09 Jul 2002 14:41:29 GMT
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Are you saying that, flying the HF in your living room corners is strictly a no no in WAF department? !!

Subject: Re: Time alignment vs reality
Posted by Paul C. on Tue, 09 Jul 2002 15:22:01 GMT
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Well, the piano, or any instrument is like Popeye... "I yam what I yam."It is not a worry about time alignment, or phase, as each key produces its sound, and it is intact, it sounds like it sounds. What we want to do is reproduce that artificially with recording gear, media, and reproducing equipment in our living rooms, and hopefully, do it well enough that it fools our ear into thinking it is the real McCoy.The piano has other issues... tuning! Did you know that the low end, middle, and top end, are not perfect octaves apart? The octaves are stretched slightly, about 3/100ths of a semitone per octave. This helps it sound in tune to the ear. That is another issue, not relevant here.There are a number of books and papers, mostly by Dr. Arthur Benade, a physicist and clarinetist, that are most interesting. Some of his mathematical analysis looks VERY much like horn speaker math!No, we can't make all the musical sounds arrive at our ears perfectly on the beat, unless it is electronically generated by synthesizers. and when music is played this way, from a midi file, it sounds very mechanical, souless. Some programs have a setting to slightly randomize the tempo, ever so subtly, and vary, for example, the piano touch, so that all the keys are not hit with precisely the same velocity (electronically, that is). This gives a more human feel to the music.And, part of the sound of instruments is also reflections, echo, reverb. When recorded dry, the tone is often dull, uninteresting. Even a slight reverb, barely heard, adds warmth to the tone.Once I was called into a studio, I thought for a recording gig. The keyboard player wanted to just sample my tone, and pay me for like 15 minutes of my time. OK, go ahead, I said. So, he sampled a few pitches. But when played back on his synthesizer, it did not sound like a sax, or a sax section. It sounded like a reed organ. Cheesy. Like the instrument patches in your sound card.There is a lot that goes to make up an instrument sound. The way a note begins ("attack"), sounds in the middle of the note ("sustain"), and ends ("release" or "decay"), the way notes are connected, the way a musician phrases, uses vibrato, dynamics (loud vs soft), phrasing, etc, all go into what makes a trumpet sound different from a soprano sax, even though both have nearly the same waveform. This is what makes Marshall Royal (lead alto with Count Basie) sound different from David Sanborn. The raw "tone" is the least of it.So, a lot of factors go into what we hear as a musical sound. What we want to do in recording is capture that as well as possible, and put it in the living room, and try to make it sound like Dave Sanborn is playing in front of you. We do not need to enhance the sound, degrade the sound, change it, we want to just reproduce it.The only place you need to worry about time alignment and frequency response is on the speaker end, we have pretty good control over all the rest of it.And I think, from my ears, horns, or at least, horns on the high frequency driver, best couple these sounds to the air in the most realistic fashion.