Subject: Advice on choosing best dsp to be programable for use with acoustical arrays
Posted by John MacBain on Sat, 29 Oct 2005 17:49:34 GMT
View Forum Message <> Reply to Message

I am looking for information and advice to help me in finding the best user friendly programmable audio grade dsp's available today or a least a clearer understanding of how to compare them and choose, based on their designs and the applications I intend to use them for, while knowing in advance the trade offs involved. I am not a programmer or a mathematician so I want to start with simple intuitive programming methods I am looking for help and support in these areas as well. I am researching the ideas and methods used to construct a high grade MIMO speaker and audio management system based on dsp technology. I want the finished unit to be able to manage approximately 8 inputs by 40 outputs, there are many new processes and functions I want to include that are not available in any commercial system controllers that I am aware of. I am also interested in finding open source code that can save me time in building some of the dsp functions, maybe there are others out there of a DIY nature, that our as I, unsatisfied with the rhetoric of some, Audio manufacturing firms claiming to have built the holy grail of sound devices, just plug in our system tunings you downloaded from us and install them into your fancy new system controller and your troubles are over, but its just not that simple. Don't get me wrong I love the idea of being able from a users point of view to call up and apply useful and relative system control programs that will help the audio tech achieve great sound for everyone at what ever event, I just happen to believe we have along way to go in making this a repeatable and consistent reality. My best measuring tools thus far have been my ears and body and they are telling me to keep trying, it ain't over till the fat lady singing, sounds great to everyone, everywhere, all the time.If you believe we can achieve far greater success then what is being accomplished today; by…. not limiting are ideas to uphold the status quo of days gone by. For example: There are better ways to build digital filters that are not based on the limitations of being only copies of there analog counter parts. In not falling for the belief that the geometry and shape of many modern vertical line arrays, are the only shapes that can propagate wave fields that can be predicted, simulated while able to, produce flat Isophasic planar style wave fronts that can supposedly be aligned beam steered where the nulls and lobs can be shaped with in limitations to our desires, where we can tune the audio system to itself and it environment, possibly tuned as well in the key to the source and so should be able to read ahead and have automatic key + pitch tracking, This ones in E minor folks, now has a whole new meaning to the audio tech. The search goes on to construct systems capable of providing audio that is complete and full in bandwidth evenly on and off axis while retaining near perfect phase response between the individual components that make up the array, with large near field sweet spots design able over large audience areas, heck maybe some day under certain situations we could say hey lets partially or completely phase cancel this part of the signal maybe only for a cued part of a musical passage and phase reinforce that part of the signal. This is complicated stuff, how far do we want go with it, maybe one day arrays can be made that won't have to be limited to ideas based on battling the relationship of the transition zone between the near field and the far field, maybe we could chose like, Hmmmm…… To far field or not to far field? That is only part of the question, for if we so chose to have a far field, what kind of far field should it be? Can we build it; I think so, better than ever before. I believe many of the answers lies in the act of un-constraining ourselves too many of problems and issues involved in some of the antiquated ideas and the conflicts that arise surrounding the nature of the relationship between traditional crossover designs (the name itself might be part of the problem, I like to think of these devices as “signal splitting processing tools”) that recombine in the geometry of there counterpart acoustical arrays. Take a moment and look at the physical nature of a hypothetical audio system and the frequency ranges involved in the wave field its able ness to produce desirable wave forms in relationships to its individual components and that of the whole array and consider them synergistically both in a physical, geometrical, frequency dependants, that should and does have a strong relationship to the algorithms that are or not controlling the individual elements of the array. To be able to produce re-constructible wave forms that can hold their integrity as they travel and disperse in a controlled and predictable fashion, comprised of multiple complex waveforms we are trying to distribute uniformed evenly across the audience area, with intent driven designed soundscapes of our choosing, that we can better control over greater or shorter physical distances in the XYZ planes, tough to do no doubt. I think we can make audio systems designed with greater respect to the source waveform, array geometry, with quality of processing, low real time latency, a good choice of filter topologies, plus a clearer understanding of the acoustical environment and its inter action with the array modally, spectrally and temporally, along with array positioning, and the use of good measurement and simulation tools involved, then we might stand a chance at resolving some if not all of these challenges. So I’m looking for, Opened minded people that have experienced sound for some time from both a scientific point of view as well as from a musical point of view and have the wisdom and common sense that only experience and passion can bring. To join forces with me by collaborating in furthering our knowledge and expertise to bring forth a era where products are evolving from self organizing systems capable of greatly improving the advancement of sound propagation theory and it applications. To contribute in architecting a better future with better sound wave control and all the possible advantages these newer systems will be able to support, for instance creating 3D acoustic holographic designed wave fields, and their synthesization, by using the proper mapping tools and system architectures. Presently I am working on a DIY project where I have designed and constructed a series 1 proto type vertical acoustical line array “The Wing Array” (still considered by me as a work in progress, it is far from perfect and has room for improvement and I need help to complete particularly the control system) consisting of; 78 separate loudspeaker elements, based on some of the previous statements and ideas I have made thou not all are necessarily contained in this document. The following is a brief description of it design and it’s limitations due to lack of expertise and the financial situations of its designer:Stereo left and right multi-way vertical loudspeaker array, (eventually I will design and build a center array etc..etc… this is a step by step endeavor)Brief description of each side of the arraysHi’s 6 hi powered ribbon drivers (60 watts rms, 2000 short peak, 120 watts rms if I ever can ever get the cooling system required to achieve this spec). Drivers are .163m in length on custom waveguides measuring .2m vertical, eff (1w@1m) 107 db spl, freq res 1khz-30khz, Imp 13 ohms, dc res 12.4ohms less than 5% @ 20 degree cel. Vertical spacing between ribbons @ .048m. Each ribbon driver is on there own amp channel presently using QSC PL 1.4. to power these elements. Over all vertical length of ribbon line is 1.89mMids12 high powered transducers (250 watts AES rating). Speakers are 166mm in diameter, Imp 16 ohms, eff (1w@1m) 97 db spl, max 117db, fs 125hz, dc res 11.6 ohm, bl 16.1, xmax +- 2.0mm. Vertical spacing between transducers @ .006m. Speakers driven two per channel, power presently provide by QSC PL 1.8.Overall vertical length of Mid line source array center to center is 2.04m. Horizontal center to center spacing between Mid and Hi line @ .22m.Lows12 high powered transducers (700 watts rms). Speakers are 304.8mm in diameter, Imp 8 ohms, eff (1w@1m) 95 db spl, fs res 39hz, V.C. diameter 101.6mm, bl 17.4, xmax 4.8mm. Vertical spacing between transducers @.045m. Speakers driven two per channel (sometimes three per ch when resources are taped), power presently being provided by Crown MA-5002VZ. Overall vertical length of low line source array center to center is 3.97m. Horizontal center to center spacing between low and mid line @ .4mSubs8 high powered transducers (1000 watts rms). Speaker are 429mm in diameter, Imp 8ohms, eff 1w@1m) 94 db spl, fs res 33hz. Speaker driven two per channel, power provided by Crown MA-5002VZ, Subs arrayed horizontally, spacing distances are played with to change modal and coupling frequencies. Don't know much more about these speakers or the enclosure design they come in other than to say they are not made by me, but manufactured by a large audio corporation. They move a lot of air, but the ways, in which the enclosures couple to structures physically and to a lesser degree modally are for the most part difficult for me to control at this time, a shame for there is such a large amount of sub base energy that is wasted by the omni directional dispersion patterns. I guess what I'm saying is, KISS this one for a little while longer, don't want to get to ahead of ourselves. I feel I have accomplished a great deal towards completing this project, I have a working sound system and thou far from perfect it's the best audio tool I've used thus far to date. It would be wonderful if I can find the help I need to realize its full potential, and then I can move on; maybe we could collaborate on a series two proto type system or some other original design ideas that are burning inside us, or that this world is asking us to consider by putting it right in our face, that we might feel a whim to create, maybe its in us. This is not a commercial venture I would like to keep this all open source, so if there are any advantages or spin offs, they are shared globally and not controlled by any one entity corporation or individual source. This does not mean that I don't think we should not give credit to ourselves and each other for contributing by the works of our labors; I'm just not convinced that the present systems in place are in our best interest. Let's quit making weapons and start making a better World for all of us to live in; we can do it I know we can.Thanks, I Hope this all makes sense there is probably a lot of mistakes in this document as it is a first draft and like me a work in progress. I have a mountain more of useful info regarding this project for those who are interested all though some is still not finished or is in rough form. CheersJohn MacBainforgetabotit@hotmail.com

---

Subject: Re: Advice on choosing best dsp to be programable for use with acoustical arrays
Posted by lcholke on Sun, 30 Oct 2005 01:39:02 GMT
View Forum Message <> Reply to Message

Hi, JohnDo not loose that enthusiasm! DSP is tough.You can also try posting over at diyaudio.
-Linc
http://www.diyaudio.com/forums/forumdisplay.php?forumid=9
