Subject: Bastanis Prometheus vs PHY - anyone heard them?
Posted by Russell Dawkins on Sat, 23 Apr 2005 02:59:18 GMT
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Has anyone had the chance to listen to both of these and offer an opinion on the sound of either / both? I am on the brink of purchasing the Prometheus MKII Airforce on the basis of what I have read so far but still would like some further assurance. I intend to be using them for mixing and mastering acoustic music, including symphonic, and so my primary requirement apart from being able to sound effortless at realistic levels on orchestra (with my Hsu subs for help) is truth of timbre. They will be replacing the Tannoy Ardens I have used for 10 years (15" 175 liter ported boxes) but now find a little tiring to listen to for long periods and less than refined through the treble (4K up).Living on an island on the west coast of Canada makes it hard for me to hear these things in the flesh, unfortunately.Thanks, Russell

Subject: Re: Bastanis Prometheus vs PHY - anyone heard them?
Posted by Duke on Wed, 27 Apr 2005 13:37:16 GMT
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Hi Russell,I've heard the Prometheus and PHY speakers (I sell the latter), and given your requirements (reproduction of symphonic music at "realistic" levels) I don't think the PHY's will work well for you. They are essentially full-range drivers augmented by a supertweeter, and have a limited maximum excursion capability. When the voice coil is driven out of its linear range non-linearities set in and the sound becomes congested. They're very nice at reasonable output levels, but can't do symphony at "realistic" levels. That's a very tall order.The Bastanis Prometheus has much more output capability, would in my opinion be the better choice of the two. The design is a dipole over much of its range which often gives a very pleasing sound, but a sound which is strongly influcenced by the room's acoustics since there is a relatively high ratio of reverberant to direct sound. Unfortunatly what can happen is that the increased room interaction, for better or worse, tends to make the differences between one recording and the next less obvious. I sell full-range electrostats which are very pleasing to listen to, but don't think that's the optimum loudspeaker format for mixing and mastering a recording.You mention truth of timbre, and in my opinion this is probably the most important thing for a loudspeaker to get right. You may have noticed that live instruments do an excellent job of giving you natural timbre even from outside the room, while few loudspeakers can do so outside the sweet spot (if even there). And you may have noticed that most loudspeakers become fatiguing after a while. The factors that go into natural timbre play a role in getting the tonal balance right over a large area, and in minimizing listening fatigue (not that you need a wide sweet spot for your application, of course).You see, the ears derive timbre not only from the first-arrival sound, but also from the reverberant sound. And because most loudspeakers have radiation patterns that vary significantly up and down the spectrum (due to driver beaming), very few loudspeakers get the reverberant field right. In most home listening rooms the reverberant field is dominant when you are six feet or more back from the speakers. If you want to check the tonal balance of the reverberant sound and see how close it matches the direct sound, step outside of the room and listen through the open doorway. Now all you can possibly hear is the reverberant sound. A live

instrument still sounds live through the open doorway, but very few speakers can give a convincing illusion of live music happening back in the room. It is my opinion that a significant discrepancy between the tonal balance of the first-arrival and reverberant sound is a significant contributor to listening fatigue (which in a professional mixing and mastering situation could take all the fun out of it). The ear/brain system is constantly analyzing sounds to see if they are "new" signals or reflections. If the reflections don't have a similar spectral content to the first arrival sound, I believe that the ear/brain system has to work harder to classify and integrate the reflections with the eventual result being a headache. Early reflections (those arriving within say 10 milliseconds or so) should be minimized, as they can have a detrimental effect on imaging. On the other hand, later-arriving reflections are beneficial, as they add a sense of spaciousness and richness to the music. Spaciousness requires multiple reflections from all directions, and if that reflected energy has the correct spectral balance (again, something live instruments get right but speakers almost never do) then timbre will also benefit. So for natural-sounding reproduction in the home, you want a lively rooom with diffusion or absorbtion of the early reflections. In a mixing/mastering situation you might want more absorption just so that you hear more of the recording and less of the room, but recognize that for pure listening pleasure the requirements are a bit different. Another factor to consider is that you don't want the tonal balance of the system changing with volume level. This can happen if the different drivers in a multiway system have differing power compression characteristics, and/or differing level-dependent distortion characteristics. You may have noticed speakers that sound dull and lifeless at low volume levels, then "come to life" when cranked up a bit, and finally become bright and even harsh at very high volume levels. Speakers using high quality prosound drivers are much less likely to have these drawbacks.Low coloration is also very important, as the less of the speaker you hear the more of the recording you hear. Coloration is minimized by starting out with a smooth frequency response free from significant resonances (aside from the fundamental bass resonance), suppressing box resonances, eliminating diffraction, and getting the radiation pattern right so that the reverberant sound sounds like the first-arrival sound.Note that diffraction is an especially nasty type of coloration. Not because its effect on the frequency response curve is all that large, but because it happens later in time than the first-arrival sound. The ears are very poor at masking an anomaly that happens at a different time from the original signal. And unfortunatly it's impossible to equalize away, because the equalized signal will still be diffracted. It has to be addressed acoustically, via attentive loudspeaker design.Now that I've described some of the factors that I think are important to your application, I've sort of painted myself into a corner; you see, I'm a dealer for a system that does the things I've described here, and site rules frown on advertising. So I'll refrain from mentioning the name here (will try to e-mail you with a link; if you don't hear from me then shoot me an e-mail). The general concepts I've described apply regardless of which specific system you go for - especially the importance of getting the reverberant field right. Only a handful of speakers do so, but in my opinion it's well worth pursuing in an application like yours. Bst of luck to you!Duke