Integrating monitoring into the chain

During the stipulation of requirements for Blue Sky's nearfield and midfield monitors, the company started at the end of the sound chain and worked backwards to the beginning. Blue Sky International co-founder PASCAL SIJEN explains the process and the integration of the subwoofer.

OVER THE LAST 25 years audio technology has changed and improved dramatically. Twenty five years ago people in their homes were listening to scratchy vinyl records and watching movies on analogue cable or VHS, in mono. The speakers they used were those that came built in to their TV, or at best, stereo 2-way ported bookshelf speakers. In their cars, they had a choice of AM radio, FM radio, cassette tapes, or 8-track tapes, which were played through 6 x 9 speakers with whizzer cones. Back then the typical 2-way studio monitor was much better than what the typical consumer had available to them for playback.

However, today's consumer now has a dizzying array of choices, such as HDTV, digital cable, satellite TV, DVDs, MP3, video games, and so on. Many of these sources are capable of high bandwidth, wide dynamic range stereo or multichannel audio. Likewise, speaker technology has also improved with the use of stereo and 5.1 speaker systems, with integrated subwoofers. Similar technology advances have occurred in the car audio world. Drivers now have access to CDs, DVDs, digital satellite and HD radio and, like the home, many autos now have complete, factory installed, stereo and 5.1 bi-amplified speakers systems, also with integrated subwoofers.

Following a parallel course, most studio equipment has improved much over the last two decades. With the introduction of digital audio workstations, hard disk recorders, digital mixers, all of which include high performance 24-bit A-DCs and D-ACs, studios today can essentially record DC to daylight. Unfortunately, studio monitors in many studios have not kept up with this trend and in many cases seem to be stuck in an infinite time loop. Like 25 years ago, many professionals are still mixing content for today's consumer systems on a set of conventional 2-way ported monitors, which tend to exhibit poor low frequency extension. Looking at this situation, we felt that there had to be a better way.

Next we moved one step up the sound chain to examine the precise reasons why a typical 2-way nearfield monitor is no longer suited to the task of creating mixes destined for today's consumer.

Almost all 2–way nearfield monitors are ported, use a 5-, 6-, or 8-inch woofer and have a low frequency cut-off between 38 and 65Hz. Since ported designs roll off at 24dB per octave, or greater, these monitors are incapable of reproducing much of anything from 20Hz up to their lower cut-off frequency. This kind of performance was perfectly acceptable when consumer playback systems had similar performance limitations. However, since many of today's consumers have full-range speaker systems with subwoofers, the typical 2-way ported design, just doesn't cut it anymore.

The other problem is what happens when you place these monitors in a typical recording studio. Although major movies have their final mix completed on a large dubbing stage, typically with a volume of 20,000 cubic feet or greater, a lot more material is mixed in studios with an internal volume closer to 3000 cubic feet, especially for music, radio and TV applications. Unfortunately, as the physical dimensions of the studio get smaller, the acoustic conditions change as well. The biggest change occurs at low frequencies, which in a large space is an issue relating to low frequency reverberation time. When you move into a smaller studio, the main acoustic factor at low frequencies is room modes, or standing waves. Room modes occur in all rooms/studios at frequencies where the wavelength of sound is an integer fraction (i.e. 1/1, 1/2, 1/3, 1/4, etc) of the distance between two walls, or the distance between the ceiling and floor.

Whenever you place a speaker in a small room or studio, its measured low frequency response will be altered by the boundary effects and room modes that form in the studio. This means that invariably some frequencies are reinforced and some frequencies are cancelled, resulting in peaks and dips in the frequency response at the listening position. These frequency variations change depending on the location of the speaker and where the listener is located in the studio relative to the speaker and boundaries of the studio. Because of this, bass reproduction from multiple speakers in a studio can be very inconsistent. To add a further complication, the speaker location that is best for imaging is almost never the best place for bass reproduction. So given little choice, recording engineers choose imaging over low frequency performance.

Granted, the use of broadband absorption, which we consider very important, can reduce the effect of the studio/monitor interaction to a degree and absorption can definitely be used to improve low frequency performance, but in a typical small studio, broadband absorption will not fully address these problems.

So what is the solution? True full-range monitoring, is the phrase that best describes our goal. Not just full-range monitoring for those willing to spend huge amounts of money on large in-wall monitoring systems, but true full-range monitoring for all applications, from the desktop on up. The technologies we employed to achieve this are based on well understood principles of physics, are relatively simple to implement and deliver superior real-world results.

First, to provide real low frequency reproduction, reduce intermodulation distortion and to reduce the influence of the studio on low frequency reproduction, we decided to incorporate a subwoofer as an integral part of the monitoring system. This is in sharp contrast to an optional subwoofer added on to an existing 'quasi full-range' 2-way monitor.

The second improvement was to eliminate ports or passive radiators and go with sealed box designs. There were three reasons for doing so. Sealed box speakers have superior transient response when compared to ported or passive radiator designs. Satellite speakers using the correct sealed box design integrate much better with a subwoofer than typical ported speakers. The 12dB per octave roll-off of a sealed box subwoofer provides a better match to the rising low frequency characteristics of small rooms/studios.

This 'room gain phenomenon', which was documented in an AES paper by Louis D Fielder of Dolby Labs, shows that smaller sealed rooms, such as the typical music studio, exhibit a 12dB per octave gain below 30 to 35Hz. This type of roll-off perfectly matches the sealed box response of our subwoofers, allowing for in-room low frequency extension down to below 20Hz. Compare this to a typical ported or passive radiator roll-off of 24dB per octave, or greater, and you can see why the sealed box response is a much better choice for accurate full-range monitoring in a typical recording studio.

The next improvement was to tie this all together with a technique called bass management or bass redirection. Bass management uses filters to extract low frequency information from two or more main channels and redirects that bass to one or more mono subwoofers. This is the same technique that is used in virtually all consumer home theatre systems and many high-end car audio systems.

Bass management when used in conjunction with satellite and subwoofer speakers provides a number of advantages. First, since the satellite speakers do not have to reproduce low frequencies they can be smaller, which makes them easier to place in the environment, and they can be placed for best imaging without worrying about how that affects their low frequency performance. Second, since bass reproduction is coming from a mono subwoofer, the sub woofer can be placed in the optimum position in the studio so as to offer the best overall low frequency response. Third, because low frequencies from multiple channels are now summed electronically, instead of acoustically in the studio, low frequency phase issues between channels are resolved in the most absolute and accurate way possible — electrically.

Now that we have a monitoring system that has the bass being reproduced by a separate source, we have to find a way to ensure repeatable and accurate setup of the system. We have found that many listeners can actually do this very effectively by ear, when using familiar broadband source material. However, to make this process a little more foolproof and repeatable, we provide a free set of .Wav test files that allow the end user to quickly adjust the electro-acoustic level of the system. These test files can be downloaded from our website (www.ablueskу.com). To use them, you just need an inexpensive SPL meter.

The purpose of calibration is to adjust the relative level of the Sub and Sat, along with the overall electro-acoustic system gain, so that 0dB VU equals a certain acoustic level at the listening position. Since most recording media is now digital, the reference electrical signal level is typically around –20dBfs with 20dB of headroom. The acoustic calibration level may vary, depending on the application and standards being used. For film applications this level is typically 85dBc, but because music is typically more compressed, a lower level, often around 78 or 79dBc, may be more appropriate.

Once the calibration procedure is completed, the end user has a system that provides extended bandwidth, seamless summation between Sat and Sub, along with an overall more accurate and repeatable system response. We believe this new methodology, which is based on simple and proven technology, makes for a clearly superior monitoring system. This true full-range monitoring system design allows the engineer to create more compelling full-range mixes that translate exceptionally well to the wide variety of modern consumer playback systems currently available on the market.

A common misconception about subwoofers

One of the more common concerns we hear is that because the bass in our systems is being reproduced by a mono subwoofer, the user fears that they may be able to perceive it as a separate, locatable source. This is actually not the case and in a correctly designed 2.1 system using proper bass-management it should not be an issue at all.

To understand why this is the case, we must first understand how our brains process location cues from our ears. Above approximately 700Hz (depending on the size of your head), your brain uses Interaural Level Difference (ILD) as the primary factor in determining the directional location of a sound (a slightly over-simplified explanation). ILD is the difference in level of a sound between your two ears.

Below approximately 700Hz, your brain begins to rely on the Interaural Time Difference (ITD) between your ears, also known as — phase shift, to determine the directional location of a sound. This works very well until the wavelengths get very long, the source becomes omnidirectional, such as a subwoofer which radiates energy spherically in its pass band, and you are in an enclosed space. In an enclosed space, such as a studio, with a source that is radiating spherically (again, such as a subwoofer), the ITD will be close to zero. This is because energy from the source is arriving at the listener from many paths, with many overlapping time differences and your brain will not be able to derive the primary location cues from your ears. Therefore your directional acuity at these low frequencies will be near zero.

However, you will have very high directional acuity at higher frequencies and because your directional cues are coming from the Sats, which typically are playing the harmonics of the LF fundamentals, this is where your brain believes the sound is coming from. Provided there is no audible distortion or sonic artefacts at higher frequencies (port noise, etc), and the sound emanating from the subwoofer is limited to below approximately 100Hz, it will be impossible for the listener to identify the location of the subwoofer in a studio.