Being 'One' With Your Source

Love, lust and getting it right

By Karl Winkler

In the April issue of *Live Sound*, I covered the three 'styles' of FOH mixing – Reinforcement; Reproduction; and Production – and how they apply to your goals as a mixer and the requirements of the gig. Now I'd like to delve further into some techniques behind achieving the best results for the first two styles, in order for your mix to be as transparent as possible. Generally, this would be most useful for acoustic music, and even perhaps for reinforcing a single speaking voice such as in a lecture or sermon.

Figure 1: Mix sound with only the best Arabica coffee in your veins.

A quick story: Several years ago, I was in Houston on a business trip. I had been in contact with Scott Fraser, the engineer with the Kronos Quartet and he invited me to come to one of their concerts since they were performing while I was in town. While there, I was really enjoying their music, which, by the way, was all over the map in terms of style. But during the concert, I kept thinking to myself that they must not be using the PA at all.

I noticed that there were microphones on the stage, and that Scott was there 'mixing', but for the life of me I was unable to pick up on to the fact that the music was amplified. During one of the pieces, however, I could hear some radical-sounding effects coming through the PA, so at that point I thought I knew what was going on.

After the concert, I talked to Scott and asked him how much of the sound was in the PA. His answer surprised me: He said that everything was in the PA, "at least a little bit". So I asked for details about how he had accomplished such a transparent mix. He indicated that first he used good microphones: Neumann KM150 small-diaphragm condensers "for their natural sound and nearly perfect polar pattern", and he also mentioned that Countryman Isomax lavalier mics installed on the instruments themselves, were used to pick up sound for effects sends.

But probably the most important factors, according to Scott, were that he made sure to keep the PA volume in line with the acoustic sound, and also that he had delayed the signal in the PA so it would follow the original acoustic wave into the hall. I was very impressed with his detailed knowledge of all these issues, but perhaps most importantly, that he had a goal regarding the effect on the audience, and carefully used the available tools to achieve that goal.

SO WHERE DO WE START?

I think it is fairly easy to lose the forest for the trees in our business. First, there is the love, er, lust of the gear. Who doesn't drool over the latest digital console with the built-in espresso maker? (See **Figure 1**) And what about the 1,000-watts per channel amplifiers in a single rack space? Or the carbon-fiber speakers which weigh 12-pounds but are 105dB @ 1W/1m sensitive with perfect coverage and directionality down to 40 Hz? Sure, these things are all cool and necessary for the advancement of our industry and craft. But I think it becomes very easy to think mostly about gear, about technical issues and getting things loud rather than the real main purpose: the music.

CRITICAL LISTENING

First of all, I always recommend that engineers spend time listening to acoustic music, in whatever form it takes. Go to jazz clubs where you can sit near a trio. Go to a symphony concert. Go see an opera. Go hear some chamber music. Check out some bluegrass. The reason for this is that despite all the amazing technical advancements we've seen over the years, there is still a huge disparity between what instruments and voices sound like acoustically, and what they sound like when pumped through a sound system.

There are two main reasons for this gap. The first reason is technical: Microphones don't hear things the way our ears do, speakers don't produce sound the way instruments do, and no electronic systems blend signals they way an acoustic space blends the collective sounds of real sources. There is only so much we can do about these things. However, it is important even to do what little we can in this area.

Start with careful mic choice (remember Scott Fraser's mention of the importance of an accurate polar pattern?) and even more importantly, mic placement.

For this last point, we would do well to understand how the sound comes out of the various instruments (Clue: Most of the sound of a saxophone, for instance, does NOT come out of the bell) Pay attention to gain structure (i.e. the official broken record subject of the audio industry. Wait, you do know what a record is, right?) This way, we can keep noise and distortion to a minimum and dynamic range and headroom maximized throughout the signal chain. And of course we must understand speaker placement and coverage beyond just that "line arrays are cool".

Another thing to remember is that fewer mics equal fewer phase problems. If you have mics all over the stage, each picking up multiple version of the same source (say, drums), you will have comb filtering problems in your output. This problem sounds 'hollow' or 'phasey'. It's a dead giveaway that you are using a sound system.

AND THEN THERE'S THE REALLY TRICKY STUFF

But beyond the general technical approach of simply doing things right in the first place, there are a number of techniques available to further the cause of making the PA disappear. The first thing has to do with the way different frequencies propagate and are absorbed.

Generally, high frequencies are more readily absorbed than mids or lows. At the same time, it is more difficult to produce low frequencies acoustically, and to manage them between the PA and the hall. The result of this is, for instance, if you had a group on stage such as a jazz band, and they played acoustically only in a theater, what you would likely hear is a dull, 'midrangey', ambient sound when sitting out in the audience.

In other words, the acoustic instruments don't generate much bass, the highs that were created on the stage got absorbed before they reached you, and

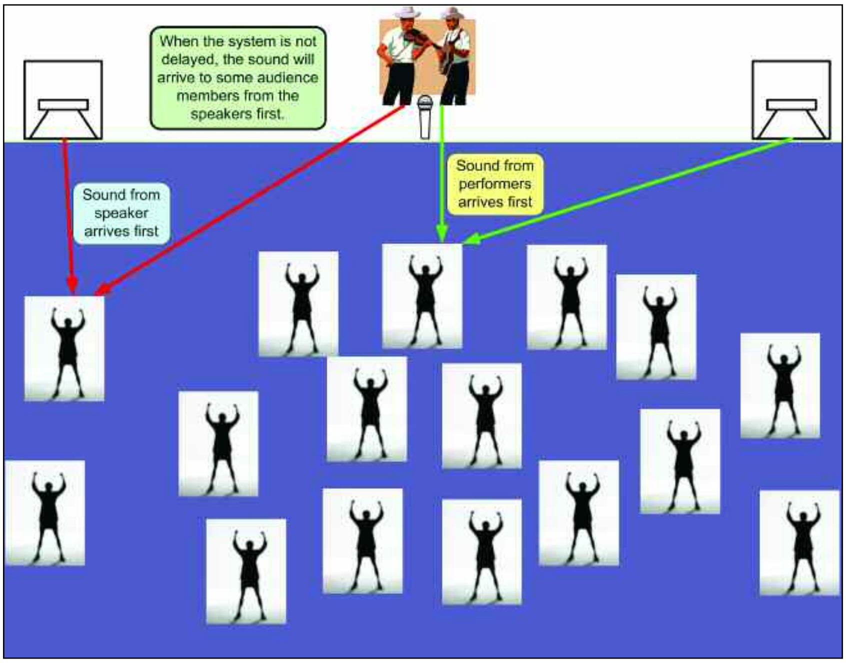

Figure 2: Notice that for listeners to the left or right of center that the sound arrives first from the speaker, then from the performers.

the reflections would be more prominent than the direct signal. In other words, it would sound mostly crappy.

So let's say we want to introduce a PA to bring the listener closer to the band, or perhaps bring the band closer to the listener. We could carefully add bass to selected instruments (such as, well, bass) and we could add highs to things like cymbals and piano. Only by doing this, and leaving everything else alone (i.e. not boosting the mids to the same degree), we might already be well on our way to improving things for the listener.

missing from the audience position would be boosted in the PA. And by having our speakers carefully pointed at the audience and away from the walls, we have introduced more direct sound to the audience to overcome the natural ambience. So far, so good.

SOUND TRAVELS AT THE SPEED OF, WELL… SOUND

One real problem with any PA mix is that now, the sound will appear to be coming from the speakers and the illusion of 'no PA' is destroyed. The reason for this is simple: The speakers are closer to the audience than the real instruments are. And since sound travels at a relatively slow 1,130 feet per second at room temperature at sea level, even fairly small differences in distance (say, 10 feet or greater) can be noticeable.

When sound from one source arrives before an exact copy of that sound from another source, the ear senses the direction of the sound as coming from the one that arrives first. This is known as the precedence or "Hass" effect. See **Figure 2**.

Now take this a step further: Imagine that you have a person giving a speech at a podium at the center of a stage, but the PA speakers are to the right and left of the stage. Audience members exactly in the center of the venue may have the illusion that the sound is coming from the speaking person, because sound from both loudspeakers will be arriving at the same time as the voice, to those listeners. But the further to the left or right that you are in the audience, the more dramatic the shift will be, so that the sound of the speaking person's voice will seem to come from the loudspeaker on that side of the stage.

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The next step might be to subtly boost those instruments that can't compete, say, upright bass (again), piano, and jazz guitar (if un-amplified). Now, we would have a balanced sound between instruments in the group, and only those frequencies

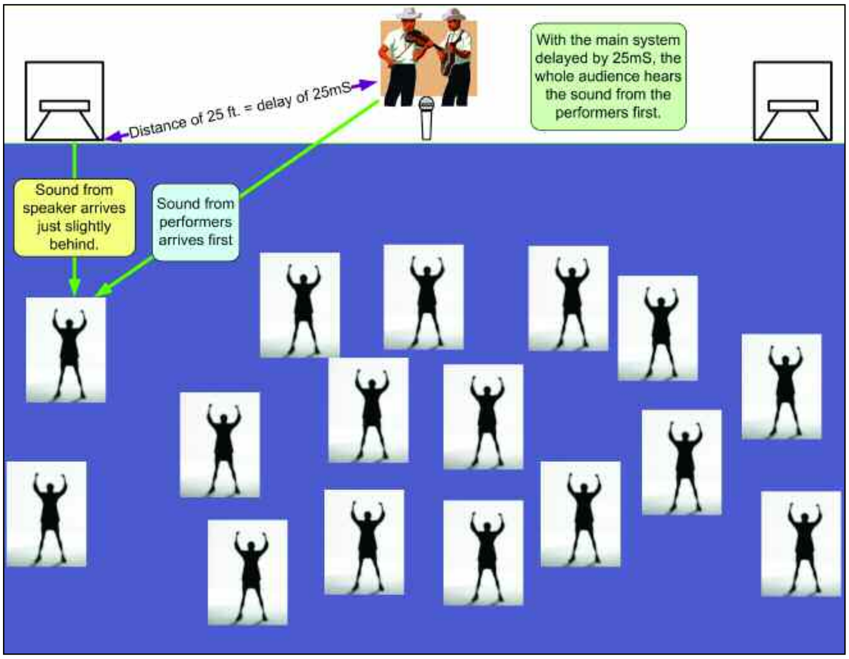

Figure 3: With a properly delayed system, the whole audience will hear the sound as if it were coming from the performers.

The first approach to fixing that problem would be to hang a center cluster. This helps to 'draw' the sound back towards the center no matter where you are the audience. But generally, even this is not enough, because the sound will still arrive to a large portion of the listeners from the loudspeakers before it arrives from the person talking. With a single person up on stage, it's not so bad. But with a whole band, you are talking about 20 to 40 feet between the back of the group and the stacks. That's up to 40 milliseconds (ms) of delay – certainly audible.

So after dealing with acoustic problems by using acoustic solutions as much as possible, you are now confronted with using electronic solutions. The digital delay is your friend in this situation, and, fortunately, these things are now cheap and plentiful. Digital mixing consoles, for example, often have them already installed on every input and output channel.

Here's the basic technique: Delay your main PA loudspeakers by the amount equivalent to the distance between your loudspeakers and the back of the band. (See **Figure 3**) You may also choose to use the drums as a reference, since it's a good idea to have percussive instruments well aligned to the PA wavefront. The ear is more readily able to identify timing differences with transient sounds, i.e. from drums. Also, drums produce quite a bit of their own acoustic sound pressure, so the audience will typically hear both the acoustic and the PA version of that sound, which is not necessarily true for most other instruments on stage.

For example, if your drum set is 30 feet behind your main stacks, then you should delay the feed to your stacks by about 30 ms. This way, as the sound from the drums comes off the stage and into the audience, the sound from the PA will line up with it. I typically add an additional 5 to 10 ms to that figure, just to ensure that the original sound arrives first. This gives the distinct illusion that the sound is coming from the band rather than the PA.

TO WRAP IT UP

All these methods may not be needed all the time. But since you are now aware of them, you can incorporate these ideas as the situation dictates and as the budget allows. And to some extent, the irony of this is that many people at the gig may not even realize that the band was amplified. Thus, you won't get any compliments, phone numbers or panties thrown at you like you usually do. But deep down, you'll know that you've achieved some magic – and that the audience experience was enhanced in a subtle but real way. After all, isn't it all about the music? OK, maybe not. I'll leave it up to you to make that choice. ■

Karl Winkler is director of business development for Lectrosonics and has worked in professional audio for more than a decade. Reach him at karl@karlwinkler.com.