Cut 'Em Off At The Pass

The effective use of high-pass filtering. *by Bruce Main*

SO THERE I WAS, SYSTEM ENGINEER at a county fair gig. The act of the day was a traveling '60s review with three or four artists who were, shall we say, past their prime. They weren't carrying engineers, so we got the duty. Soundcheck went fine. The artists cruised through their paces and the hired back-up band was surprisingly good. Nothing to do but hit catering and wait for the "white hair, blue hair and no hair" crowd to show up.

Show time. The band started the intro, everything was rocking in an old school sort of way and the emcee/star came out. He was much more animated than he had been at soundcheck – running around the stage, exhorting the crowd to put down their walkers and dance, generally getting them in the mood.

Suddenly I heard a phantom kick drum that was waaaay off the beat. I cued up my cans and began to solo channels. The offending thump came and went, but I finally put my eyes and ears together and realized that the star, we'll call him "Frankie" for the sake of this article, was running around clapping his hands while holding his SM58.

At first I tried riding the mute button on his microphone, but I was spending so much time on him I couldn't mix the rest of the show. So I reached for the variable high-pass filter knob and ran it up to 300 Hz. It thinned his voice out a bit but I doubt anyone noticed but me. Problem solved.

The earlier we can deal with these issues in the signal chain, the better.

COMBAT THE UNWANTED

High-pass filters are probably one of the most under-utilized features on the console. The most common use has traditionally been to combat unwanted proximity effect, which is the tendency of directional mics to increase their output at low frequencies as the sound source gets closer to the mic.

Cardioid and hypercardioid mics get their directional characteristics from ports in the mic capsule that allow sound to impinge on the rear of the diaphragm as well as the front. The added length of the ports creates a difference in path length between sounds hitting the front of the diaphragm and the rear. Pressure differences between the front and rear of the diaphragm are what make it move. These different path lengths cause a difference in pressure because of two factors: phase and amplitude.

The phase component is dominant at higher frequencies. A 20 kHz wave is slightly more than a half-inch long. The path length difference from the front of the diaphragm to the rear is large as a percentage of the wavelength, so almost complete cancellation can occur. This is one reason why microphone directivity breaks down as frequency decreases, and it is also why the diaphragms of cardioid mics are damped at about 6 dB per octave as the frequency rises. Remember: more pressure difference equals more diaphragm movement.

But the key to proximity effect is the amplitude disparity. The inverse square law tells us that every time we double the distance from the source to the diaphragm, we lose 6 dB. This is very powerful at short distances; for example, the difference between a singer being a quarter-inch from the mic and a half-inch from the mic is 6 dB.

It also means that the difference in path length from the front of the diaphragm to the rear becomes more and more significant as the source gets closer. Since phase cancellations are a fixed percentage of amplitude at any given frequency the amplitude factor becomes much more dominant at close distances than the phase factor. The phase part of the equation has less and less effect at longer wavelengths while the amplitude part holds true at all frequencies. Hence proximity effect.

Proximity effect can go as high in frequency as 500 Hz depending on the mic, although 200-300 Hz is more common. The amplitude gain can be as much as 16 dB! This is probably why high-pass filters were put on mics and into consoles in the first place.

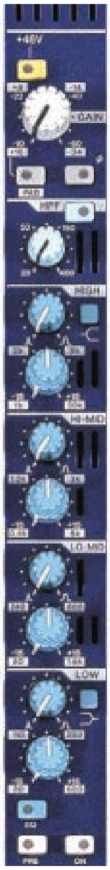
ANOTHER BENEFIT

But sweepable high-pass filters can also be used to help you clean up your overall mix. One of the things we learn from audiology is that lower frequency sounds obscure higher frequency sounds, but not the other way around. This is one of the principles that makes sound masking work. It's useful in sound masking systems, but in a live performance situation, not so much.

Many live mixers react to this unconsciously when they reach for the house graphic and hack away at 125 Hz and 160 Hz. True, many rooms react poorly in that frequency range, but the room is only one part of the problem. Let's think about the physics of low frequency sound waves.

A 100 Hz wavelength is 11.3 feet long (at sea level, at 72 degrees Fahrenheit, etc., etc.). This is typically above the crossover point for subwoofers, so it's probably being reproduced by the main arrays. In order to provide good directivity at any frequency, the array must be larger than the wavelength. If the array is not larger than the frequency of interest, the sound waves wrap around the array and it behaves as an omnidirectional source.

Even if the line array is fairly long, you only get the directivity benefits in the vertical axis. Chances are, the array is four feet wide (or less), which means that in the horizontal plane, pattern control starts to break down at around 250-300 Hz. What is in close proximity to the array on the horizontal axis? The stage. And the mics on the stage.



Even if the subs are being run from an aux send on the console (which I highly recommend), there is still energy from the sources being routed to the subs that finds its way back into the stage mics. Because the same laws of physics hold true for stage sources as for main arrays, the mics are picking up the desired musical content in these frequency ranges – plus the adjacent instruments and floor wedges, plus the room resonances, plus the wraparound from the main system in the longer wavelength frequencies below about 300 Hz.

This is happening even if we don't consider the artist clapping with a mic in his hands or tapping his foot on the mic stand base. And to compound the problem, the cardioid pattern of the mics breaks down in the lower frequencies as well. The inverse square law (minus channel compression) is your only friend at this point!

INCREMENTAL STEPS

So, what's a poor sound engineer to do? Directional cardioid subs and cardioid sub arrays can help enormously with the

least directional part of the mains, which is often closest to the stage. We've already made gains in cleaning up the stage sound (at least in some cases) with tools like in-ear monitoring and instrument amplifiers located off stage in isolation cabinets. While these techniques are incrementally helpful, there's another tool at our disposal: the console channel's variable high-pass filter.

The earlier we can deal with these issues in the signal chain, the better, which is why high-pass filters are found on many outboard mic preamps as well. If your mics have a shelving filter, try that first. If it doesn't degrade the instrument sound, leave it switched in.

Next, at soundcheck, start your equalization process for each mic by sweeping the high-pass filter up until you hear it affect the sound. Obviously, there are some inputs that might be left out of this process like the kick drum, bass guitar and a low piano mic. DIs and other direct feeds don't count because they aren't picking up ambient sound. A sharper knee and a steeper slope will allow you to set the filter to a higher frequency without degrading the natural tone of the source up to a point. Too steep of a slope can cause a filter to "ring." Filters have resonances too.

Then, during the show, solo each mic with headphones that provide good low frequency isolation and response (I like Beyerdynamic DT770s), and you may find you can cheat your high-pass filters up in frequency a little higher. Oh and by the way, have the monitor engineer try this too, only he/she can be quite a bit more aggressive with it. The performers on stage don't have high-pass filters on their IEMs, and many ear molds don't do a great job of isolating lower frequencies.

THE PAYOFF

Using this approach should lead to cutting less in the 125-200 Hz range on the system EQ because you are solving the problem at the source.

You'll also be surprised at the increased clarity in your overall mix. The system will have more headroom as well since the frequency ranges we're dealing with are real energy hogs.

Remember, garbage in garbage out. Why deal with it in your mix when you can cut it off at the pass? The high pass, that is. ■

Bruce Main *has been a Systems Engineer and Front of House Mixer for more than 30 years, and has also built, owned and operated recording studios and designed and installed sound systems. Check out more of his writing on ProSoundWeb.com (Search: "Bruce Main").*

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