by Dennis A. Bohn

Crossovers, Equalizers & Compressors

Re-Thinking Reality And Rationally Reconsidering Processing

There are other Mojo series products, not directly associated with day-to-day sound reinforcement application such as the MH 4 Stereo Headphone Amplifier not detailed in this dispatch.

The notion of simplifying signal processing is a departure from most current signal processor designs. At present, it seems, signal processing must have at least one 12AX7 tube to be popular, and be packaged into some sort of retro WWII look, complete with big "techno-erotic"(1) knobs and meters.

Appropriate Scale

While all this is fine, so too is the idea of making your life simpler, your performance better and your budget lower.

Most musicians and many beginning engineers complain that all signal processing was too expensive and **way** too confusing when you first start out.

Predictably, many Rane end-users fitting this profile have asked for easier and more affordable signal processing, without sacrificing quality and reliability. The direct engineering response to requests, Rane's new Mojo Series of signal processing, is now showing up in live sound venues worldwide.

Mojo Series' original target market was previously unserved by Rane. The goal was to produce a line of low-cost, quality product choices which would allow users to survive/thrive and generate enough income to move up the equipment chain.

Eventually, other established players soon discovered distinct advantages to Mojo Series' functional approach to essential signal processing, offering just the essence, without frills, or confusion, and are now using it in touring systems.

Beyond mics, mixers, amps and speakers, even the most basic sound systems also require active crossovers, equalizers and compressors. By reviewing these basic processing functions, we re-evaluated processing's essence and to fully understand how such "essential" signal processing can be made better, easier, and cheaper.

The resulting Mojo methodology was to strip signal processing down to its absolute core. Go back to basics. Offer only the minimum of what's needed, but still offer the best. Strip every electronic circuit down to its minimal implementation. And if that doesn't work, create new ways of doing old things.

What follows are the details of this engineering-for-essence approach on a processor-by-processor basis.

Stereo Active Crossovers

What do you need as a minimum from an active crossover? Just two things: let you set the crossover frequency, and let you balance different driver levels (because high frequency (HF) drivers are more efficient than low frequency (LF) drivers). And with simple systems, using the same brand cabinets on both sides of the stage, you only need one set of stereo-ganged controls to achieve these functions.

Ganged controls accomplish three important things:

1.) They speed and simplify setup;

2.) They improve the performance by guaranteeing that both sides are set the same and equally matched;

3.) They save money.

Therefore, the Mojo Series MX 22 Stereo 2-way Crossover has just one frequency selection control that sets both channels to exactly (within 2%) the same frequency (more on how this is done later). Likewise, the MX 22 has only one set of controls for setting both channel's Low Out levels, and another single control for setting both channel's High Out levels.

With a single set of controls to worry about, both channels remain perfectly matched and set exactly the same. The Mojo Series offers the MX 23 Stereo 3-way Crossover with identical simplified controls.

Even VCAs?

More savings and performance improvements were accomplished by incorporating the industry's first use of advanced VCA (voltage controlled amplifier) frequency selection. This represents a brand new solution to the ornery problem of how to offer precision 24dB/octave, 4th-order, Linkwitz-Riley performance on a budget.

VCA control not only makes advanced crossovers affordable, but it gives them "truth in front panel settings" something Rane has always stressed. The diffi-

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culty in designing low-cost Linkwitz-Riley is making the crossover frequency adjustable.

Four things must change simultaneously, and by exactly the same amount. To do this accurately, with minimum parts, requires a precision (read expensive) 4-section potentiometer. Many competitor designs use inexpensive 4-section pots that do not match or track each other nearly good enough. This results in serious amplitude and frequency errors. It saves money, but it degrades performance beyond any levels that Rane considers acceptable.

Another approach eliminates the expensive 4-gang pot altogether by substituting electronic switches and logic circuits, but these designs drastically increase the component count and the manufacturing time, thus driving up the cost performance, but so does the cost. Therefore a new design was needed. Taking advantage of the latest VCA technology, a solution was found that combines very high performance with low cost.

Stereo Graphic Equalizers

Examining standard graphic equalizers to determine their main cost-defining element quickly leads to the slide potentiometer controls (i.e. sliders). With as many as 62 sliders showing up in some stereo designs, they add up to a substantial sum.

So, we asked ourselves, how about eliminating a whole bunch of them like, say, half? Applying the same reasoning used in defining the Mojo crossovers, Rane researched the question: Can you gang together the two channels?

The interesting answer was that, not only were there applications where you could use stereo-ganged controls, but that they would often work better. Thus was born the Mojo Series MQ 302 Stereo 1/3-Octave Equalizer featuring 30 stereo-ganged sliders controlling 60 constant-Q filters at 30 per channel.

So, where do you use them? Well, take budget loudspeakers, for instance. An unfortunate truth regarding budget loud-

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Passive vs. Active Crossovers

Once you start needing multiple cabinets, active crossovers become necessary. To get good coverage of like frequencies, you want to stack like-drivers. This prevents using passive boxes since each one contains (at least) a HF driver and a LF driver. Some smaller systerns combine active and passive boxes.

Even within a single cabinet it is common to find an active crossover used to separate the low- and mid-frequency drives, while a built-in passive network is used for the high-frequency driver. This is particularly common for super tweeters operating over the last audio octave. At the other end, an active crossover often is used to add a subwoofer to a passive 2-way systern.

All combinations are used, but each time a passive crossover shows up, it comes with baggage. One of these bags contains power loss. Passive networks waste valuable power. That extra power that you need to make the drivers louder, instead boils off of the components and comes out of the box as heat not sound. Ergo, passive units make you bring a lot bigger amp to the party than you want to carry.

A couple of additional problems packed with passive networks has to do with impedance. For passive networks to work exactly right, the driving impedance (the amplifier's output plus the wiring impedance) must be as close to zero as possible and not frequency-dependent, and the load impedance (the loudspeaker's characteristics) must be fixed and not frequency-dependent (sorry not in this universe; only on Star Trek). Since these things are not possible, the passive network must be (at best), a simplified and compromised solution to a very complex problem.

Consequently, the crossover's behavior changes with frequency not something you want for a good sounding system. All this, and not to mention that it is almost impossible to get high-quality, steep slopes passively, so the response suffers.

One last thing to make matters worse is back-electromotive force (back-emf): literally, back-voltage, which further contributes to poor sounding speaker systems. This is the phenomena where, after the signal stops, the speaker cone continues moving, causing the voice coil to move through the magnetic field (now acting like a microphone), creating a new voltage trying to drive the cable back to the amplifier's output!

If the speaker is allowed to do this, the cone flops around like a dying fish. It does NOT sound good. The only way to stop back-emf is to make the loudspeaker "see" a dead short, i.e., zero ohms looking backward, or as close to it as possible something that's not gonna happen with a passive network slung between it and the power amp.

Active crossovers cure many ills of the passive systems. Since the crossover filters themselves are safely tucked away inside their own box, away from the driving and loading impedance problems plaguing passive units, they can be made to operate in an almost mathematically perfect manner.

Extremely steep, smooth and well-behaved crossover slopes are easily achieved by active circuitry. There are no amplifier power loss problems, since active circuits operate from their own low voltage power supplies. And with the inefficiencies of the passive network removed, the power amps more easily achieve the loudness levels you want. lastly, loudspeaker jitters and tremors caused by inadequately damped back-emf all but disappear once the passive network is removed.

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speakers is they usually don't sound very good. This is often due to an uneven frequency response, or more correctly stated, a non-flat power response.

A conceptually ideal loudspeaker system has a flat power response. This means that if you pick, say, 1 kHz as a reference signal, use it to drive the speaker with exactly one watt, measure the loudness, and sweep the generator over the speaker's entire frequency range, all frequencies will measure equally loud.

Realistically "conceptually perfect" is usually unavailable and even the most expensive speaker systems often require equalization to level the system's frequency response.

The attack time is the amount of time that passes between the moment the input signal exceeds the threshold and the moment that the gain is actually reduced.

This is where equalizers can help overcome such response deficiencies(2). By cutting filtering a little here, and (less frequently) boosting a little there you can establish a very acceptable power response and an improved sounding system. (Editor's note: Remember: No equalizer — not even Rane — will actually any improve loudspeaker systems when high multiples of +12 dB frequency boosts are applied.)

And since you need to make the identical equalizer adjustments to each loudspeaker, stereo-ganged controls are perfect. Not only perfect, but better, because you are guaranteed that both channels are set the same, and you do it faster, and it saves you money.

The growing popularity of stereo earworn monitors creates a perfect application for stereo-ganged equalizers. Having just one set of sliders lets you quickly, easily, and most importantly, accurately adjust the timbre of each in-ear phone for your ears and your tastes.

All without the worry and tediousness of painstakingly setting each channel exactly the same. For if one ear's equalization differs from the other, by even a small magnitude difference, combined with the associated phase shift, it alters the spectral balance of the material sometimes drastically. (Name-dropping time: B.B. King uses an MQ 302 Stereo Equalizer for his in-ear stereo monitors, and so do all members of the Neville Brothers that use individual ear-worn monitors.)

Dual Channel Compressors

Compressors, of course, are signal processing devices used to reduce (compress) the dynamic range of the signal passing through them. You must reduce the dynamic range because the extreme ranges of audio are near impossible for any sound system to handle.

If you turn it up as loud as you want for the average signals, then along comes these huge musical peaks, which are vital to the punch and drama of the music, yet are way too large for the power amps and loudspeakers to handle. Either the power amps clip, or the loudspeakers bottom out, or both and the system sounds terrible.

Or going the other way, if you set the system gain to prevent these overload occurrences, then when things get nice and quiet, and the vocals drop real low, nobody can hear a thing. It's always something. So you need a compressor.

To simplify a compressor to its essence, the Mojo Series MC 22 Dual Compressor offers (beyond a standard input level control) only a threshold control and a ratio knob. Using it is simple: Set a threshold point, above which everything will be turned down a certain amount, and then select a ratio defining just how much a "certain amount" is.

All audio below the threshold point is unaffected and all audio above this point is compressed by the ratio amount(3). Feature-burdened (re: gonna-get-ya-into-trouble) compressors include attack and release controls.

The attack time is the amount of time that passes between the moment the input signal exceeds the threshold and the moment that the gain is actually reduced. The release time is just the opposite the amount of time that passes between the moment the input signal drops below the threshold and the moment that the gain is restored.

These controls are very difficult to set, and yet, once set, rarely need changing. Because of this difficulty, and the terrible sounding consequences of wrong settings, the MC 22 includes circuitry that makes these adjustments automatically, based on the program material for a wide variety of music and speech one less thing for you to worry about, or pay for.

Unlike its Mojo Series cousins, the MC 22 does not use stereo-ganged controls. This gives it more versatility. It does, however, feature a stereo linking switch. Activating this switch creates an "or-ed" situation, where whenever either channel goes into compression the other channel is automatically, identically compressed. This preserves the stereo image, which otherwise would be "pulled" to one side or the other by the compression. Since system overload is not the only place we

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find compressors, the MC 22's dual channel approach proves to be the most useful and cost-effective way.

Compressors as Sound Shapers

The other popular compressor use is in the making of sound. And here, you normally use single channels of compression at a time.

For example when used in conjunction with microphones and musical instrument pick-ups, compressors help determine the final timbre by selectively compressing specific frequencies and waveforms. Common examples are fattening drum sounds, increasing guitar sustain, vocal smoothing, and bringing up specific sounds out of the mix, etc.

The MC 22 also includes a special control-less expander. Expanders are signal processing units used to increase (expand) the dynamic range of the signal passing through it. However, modern expanders operate only below the set threshold point, that is, they operate only on low-level audio. Operating in this manner they make the quiet parts quieter.

The term downward expander or downward expansion evolved to describe this type of application. The most common use is noise reduction. For example, say, an expander's threshold level is set to be just below the quietest vocal level being recorded, and the ratio control is set for 2:l.

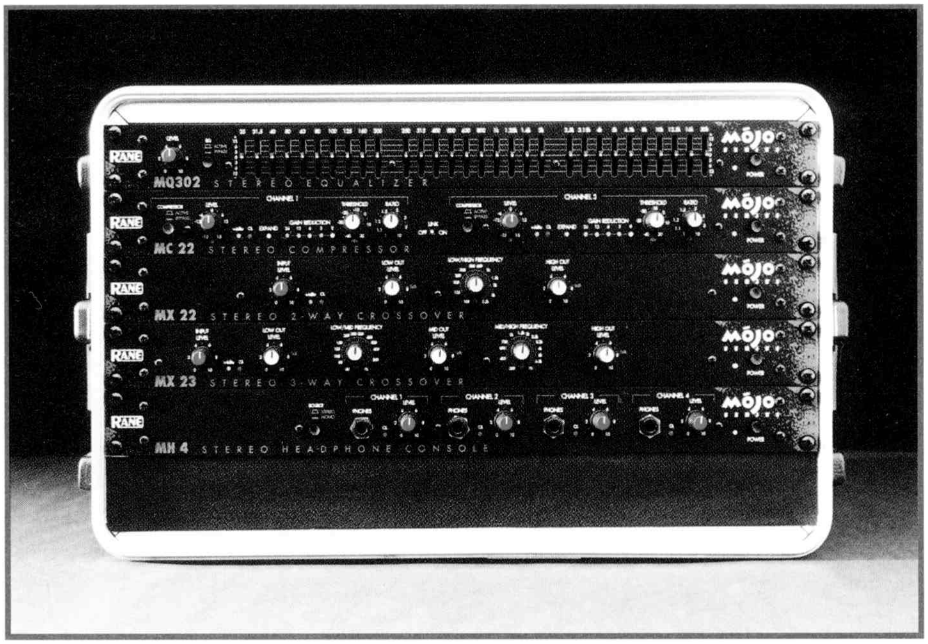

*Photo #1: **Makin Mojo Outta the Box***

What happens is this: when the vocals stop, the signal level drops below the set point down to the noise floor. This is seen by the expander circuitry as a step decrease from the smallest signal level down to the noise floor.

If that step change is, say, -10 dB, then the expander's output will attenuate 20 dB (due to the 2:1 ratio, the 10 dB decrease becomes a 20 dB decrease). This results in a noise reduction of 10 dB, that is, it is now 10 dB quieter than it would have been without the expander. Rane's *adx* circuit in the MC 22 is an automatic downward expander giving you 10 dB of added noise reduction.

So, there you have it: Rane's Mojo Series signal processing's soul. ❑

Footnotes:

1.) Thanks to Fred Ampel, Technology Visions, for this wonderful term.

2.) To determine the loudspeaker's actual response, do the measurements outside (no reflections off walls or ceiling) and up in the air (no reflections off the ground). This gives you a very accurate picture of just the loudspeaker's response, free from room effects.

3.) The key to understanding compressors is to always think in terms of increasing level changes in dB above the threshold point. A compressor makes these increases smaller For example, say the ratio control is set at "1.6." This means that for every 1.6 dB increase above the thresholdpoint the output will only increase 1 dB. In this regard compressors make loud sounds quieter That is, if the sound gets louder by 1.6 dB but the output only increases by 1 dB, then the loud sound has been made quieter It's an important concept.

Dennis A. Bohn is V.P. Research & Development at Rane Corporation.