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WHITHER TO MOVE?

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Each time we get a request to provide moving fader automation for live mixing consoles, it rekindles an old debate about the suitability of motorized faders for live sound. Moving fader automation is widely used in the recording industry because of its apparent ability to replicate operator moves. But live sound control has its own unique requirements. At the console, the live show and the recording studio could not be more different.

AUTOMATED MULTI-TRACK MIXING

In multi-track mixing, many hours are spent prior to mixdown assembling a large number of sounds to be combined in different proportions at different times during the final mix. These sounds are independently synchronized so they always play back at the same speed, in the same time relationship and at a consistent volume during the mixing process.

The process of creating (or programming) an automated multitrack mix, with or without moving faders, is one of making repeated and increasingly minute adjustments to the volume levels of all components of the sound track. This incremental adjustment process is often performed hundreds of times to produce a single track, with all sounds being recued and played back while careful attention is focused on adjusting short segments of each sound, one at a time. All the sounds not under immediate scrutiny are automatically controlled in level according to previously 'learned' mix data.

In the end, the final mix is completely controlled by the automation system - without any human intervention. This requires many hours to create a very short sound track, resulting in a painstakingly produced product.

LIVE SOUND CONTROL

The live mixing process is different from studio automation in almost every way. Although actors, singers and musicians do their best to perform at a consistent pace, relationship and volume level, there are naturally small differences from time to time which cannot be predicted or corrected with programmed level changes. In addition, there are physical variations which cause the sound to be different from moment to moment and from performance to performance. Typical variations include:

- microphone placement (distance from instrument or voice),

- microphone condition (weak batteries, obstructed or defective capsules, a substitute unit, broken antenna etc.),

- atmospheric conditions like temperature and humidity, and

- size and distribution of the audience in the house.

Because of these, we still need a human to provide intelligent live compensation. Sound design has not been taken over by 'artificial intelligence' - yet. Even if automation controls a live mix, you will still want to override or augment the programmed levels to adjust for these variations. You could employ a wide variety of correctional responses, including:

- Increasing the overall volume of the entire mix if the audience is larger or environment noisier than normal;

- Increasing an individual microphone's level for the entire show if a singer has voice trouble or if their microphone is weaker than normal;

- Adjusting the microphone each time a microphone or shared channel is swapped between an ailing singer and a performer singing normally;

- Making adjustments continuously throughout the show if the microphone's output is sporadic because of contamination, weak batteries or unstable placement.

In short, you need a system which controls mixes as programmed, but also allows the levels to be manually adjusted at any time. This requires a fader with two simultaneous but conflicting capabilities:

- automatic level control; and

- immediate manual override.

Initially, a motorized fader seems to elegantly provide both features, but this is not the case. If you wish to override the programmed position, the motor must be disabled so it will not 'fight' your hand or make operation difficult. This means all control in this channel now resides with you, the operator. To properly mix the show, you must now remember all the programmed fader moves, and duplicate them precisely while compensating with live corrections - a tall order at best. The problem is compounded by the number of channels needing correction. If you go back to programmed control you can no longer override or modify the 'canned' settings.

This would all be easier if the control system allowed setting an offset level above or below the level initially programmed. It would also be useful to vary the offset up and down. In fact, if we simply put a standard, non-motorized audio fader beside each moving fader and have the audio pass through both, we meet these requirements without a complex control system.

Now, with side-by-side faders both adjusting each channel's volume level, you control the standard fader and the automation system controls the moving fader. You have no need to touch the moving fader since it is motorized and because any live volume adjustment can be easily obtained with the regular fader. Manually adjusting the moving fader may be useful in programming automated moves, but it is in fact technically easier to use a regular fader to copy moves and transpose them to the moving fader.

So now we have an environment meeting all operational needs, but every audio channel incorporates an expensive moving fader which, oddly enough, you never need to touch. Why provide the second, moving fader? Could we not simply replace it with a variable gain element such as a voltage controlled amplifier (VCA) or digitally controlled attenuator (DCA)? In fact, we could simply integrate an automation system with our existing console since programming does not have to be done by the console. This configuration has the added feature of remote or off-line programming, without necessarily requiring time in the theatre.

The automation system controls the variable gain element, providing consistent and reliable programmed volume level changes, while the fader in each channel allows you to adjust the same volume level simultaneously. The net volume level is the combination of the settings of the variable gain element and the fader. The nominal fader position is "0 dB," representing exactly the same channel gain as when the show was programmed.

You always know the volume is the same as programmed whenever the fader is at "0 dB." To compensate for a weak microphone, the fader can be easily increased as required and the show will run normally until another adjustment is needed. If the microphone is repaired or replaced during the show, simply reset the fader to the nominal position. If an understudy sings differently than the regular singer, you can continuously adjust to compensate for the different styles - all the time knowing via the fader position exactly how far the level is from normal.

This is far more logical than using motorized faders. To take over live control using moving faders, you must disable the automation then attempt to simulate all programmed moves along with all desired changes. To put the system back under automated control, you must rematch levels perfectly with the programmed settings and then enable the automation. All in all, moving faders end up looking much less attractive for live sound than a programmable control system properly integrated with a regular console.

PROGRAMMING LIVE SOUND AUTOMATION

Since automation for moving faders is based on copying the operator's motions, it could appear functionally superior for certain applications. But you must always program these systems in real time. For example, it takes at least 90 seconds to program a 90 count fade. And it would take at least a minute to subsequently change it to 60 counts - with many systems actually requiring you to reprogram the full 90 seconds. A more theatrical way to program these examples is to first enter '90' as the fade rate, then later change the figure to '60.' This process takes only a few seconds - helpful, since fades are frequently changed before the right values are found.

Recording engineers readily acknowledge it takes a long time to mix using a typical automated mixing system. Pity the poor sound designer who, after receiving three pages of tech notes at the end of a long rehearsal, has to then program new levels and fades in real time. Moreover, pity the poor sound designer with a studio-style automation system who is expected to change a number of complex programmed sound moves while the entire cast and crew wait around. People and equipment have often been removed from productions for such costly delays.

Much as in lighting control, what matters most are practical considerations - programming speed and efficiency. You don't see lighting boards designed like automated mixing consoles.

AUTOMATED VS. LIVE TIMING AND FADES

Studio automation systems require SMPTE time code - a logical convenience since every recording is a specific length and tempo.

The start of the recording is given an arbitrary 'start time' and the time code clock simply counts up from there in fractions of a second, or 'frames.' The time code clock is initially recorded adjacent to the sound elements onto a time code track. Thus, each sound element in the recording is always synchronized with an exact time - or frame - whenever it is played. Automation is always referenced to time code and an event's 'location' is identified by its corresponding frame.

The setting of each fader or gain element is defined by the system, frame by frame. The volume of each sound event in a recording can therefore be uniquely related to each event's frame time. Moreover, time code is predictable, with its frame times ascending in order and at a constant rate. This means the automation system can anticipate its next moves - 'knowing' they should always happen exactly as programmed.

But in the theatre, the timing of each scene, or even each line, can vary considerably from one performance to the next. It's always been hard to get actors to follow SMPTE time code. The most important job you have is to make the sound follow the show.

A typical need is to make programmed fades happen at the right time - and end at the right time. This involves telling the automation system to begin and end fades at other than programmed times. It takes a very special kind of automation system to start a fade, then modify it because of a need to change the fade rate - all the while constantly maintaining a continuous, smoothly changing audio level in all affected channels.

This is a key difficulty encountered when moving faders and studio style automation systems are adapted to live applications.

A system locked to time code cannot have these capabilities. Even if it could respond to varying or discontinuous time code, it certainly would not have the intelligence to respond in a predictably smooth and consistent manner.

Live performances are unpredictable and inconsistent, so a system designed for live control has to be more flexible and intelligent. For example, if you make a cue which tells an intelligent live programmable fader or gain controlling element to change 5dB in 25 seconds, it will fade smoothly at a rate of one decibel every 5 seconds. If the pace of the show changes, requiring the cue to complete sooner or later than the prescribed 25 seconds, you can instruct such a fader to go faster or slower at any time during the fade and its response will be precisely as desired. You can even command such intelligent faders to reverse fade directions at any time without unpredictable or undesirable results. Moreover, you can create cues which dynamically alter hundreds of intelligent faders simultaneously. You can execute these cues at will and overlay them as desired to produce virtually any effect - exactly as programmed and without the uncertainty and inconsistency caused by an operator attempting to do too many things at once.

Does this scenario sound too good to be true? I would be the first to admit the most elegant solution is yet to come - but it's far closer than it was a few years ago. Since today's equipment so far exceeds the performance of the past - and at a lower cost - it doesn't make sense to use any less than the best for live sound control.
