The Feedback Loop

JOHN A. McCULLOCH

● Designs and ideas on console construction are many, and while there is a basic console plan, any design which permits the flexibility desired for the particular application without becoming unnecessarily complex is satisfactory. At present a console is being constructed *not* primarily for use in recording mixing, but rather for the investigation of miking techniques, and the microphones themselves. In order to provide maximum flexibility and usefulness, expansions of some circuits not usually found on general consoles were incorporated into this unit. Perhaps the circuits as used in this console are not directly usable, but could be adapted to your particular design.

Since the module unit chosen for the input channels has separate high- and low-level inputs (selected by a switch), the additional wiring required was utilized to perform other functions. Each input pair contains a phase-reversing switch, providing control of microphone and high-level phases when mixing multiple sources into a single channel.

After the phasing networks there is a dpdt switch to place the high-level pair in parallel with the nomal microphone pair. Insofar as the combining is after the phasing, the application not only permits the addition of two microphones before any amplification, but they may be combined in-phase, out-of-phase, or in-phase themselves but out-phased with respect to the remainder of the microphones used in the pickup.

A fourth switch would permit the selection of the combining action to be either in parallel as shown in FIGURE 1, or in series as shown in FIGURE 2. Continuing along the input circuitry—a pair of test points dropped from the microphone, just before amplification, are brought up to a panel. The purpose here, is to provide a means for measuring, with a calibrated, wide-range dB meter, the actual output level of the microphone. The meter should have calibration means for the standard impedances used in microphones as well as the normal 600 ohms. Some manufacturers of amplification equipment claim a margin of 10 dB between normal operating level and maximum capabilities of the amplifier (both input and output levels should be considered) to be sufficient. I prefer to plan for a minimum of 16 dB, and several recording engineers, whose work is constantly at the forefront, claim a requirement of from 24 to 26 dB.

Whichever margin you feel sufficient for your operations, the use of these test points, the wide-range meter and an input pad (either in-console or in-line plug-in type) make it possible to accurately control the input level to the first stage of the amplifier to obtain the best possible conditions for high-quality recording.

Test points at other stages might be considered, but if the design of the console was correct and conservative, the input is the main variable. Certainly it is the most critical. We desire the highest input level possible to avoid a poor signal-to-noise ratio. Yet, we must provide a good peak margin to prevent any possibility of overload. Measure and know the knee of distortion for your particular input stages (the point on the distortion *versus* power curve where the distortion first starts to rise rapidly). Use this point, not the actual point of observed clipping, as the maximum input to the amplifier. Subtracting your peak margin will give the normal operational input level.

It is common to provide echo feeds both before the fader (pre-echo) and after the fader (post-echo). The question arises: should this post-echo be tapped directly after the fader, or after equalization? Perhaps a third name-designated position (eq-echo?) should be adopted to clarify just where the tap *is* made. The versatility of all three positions should be evident, but the complexity of switching may not be to your taste or pocketbook.

At the output of the unit or channel, we normally find buss, submaster, or master channel selection. Bussing or

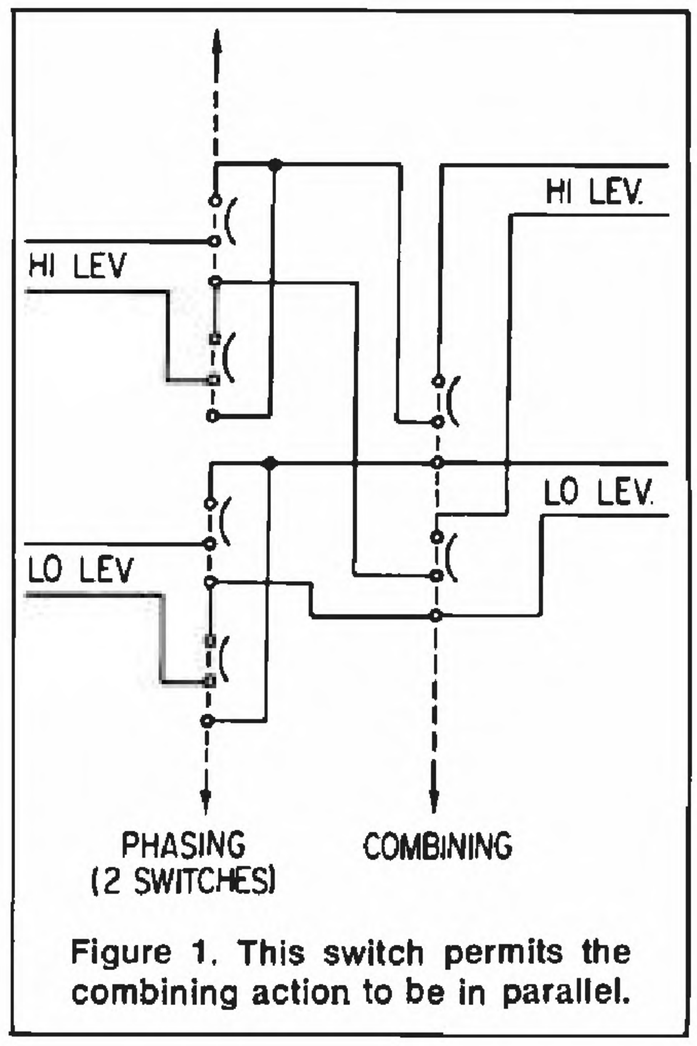

Figure 1. This switch permits the combining action to be in parallel.

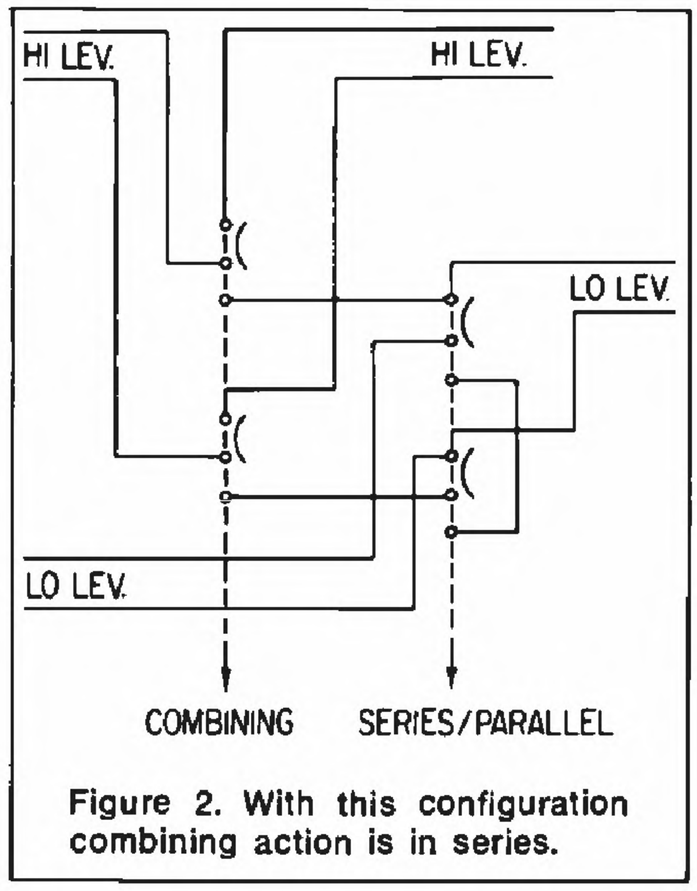

Figure 2. With this configuration combining action is in series.

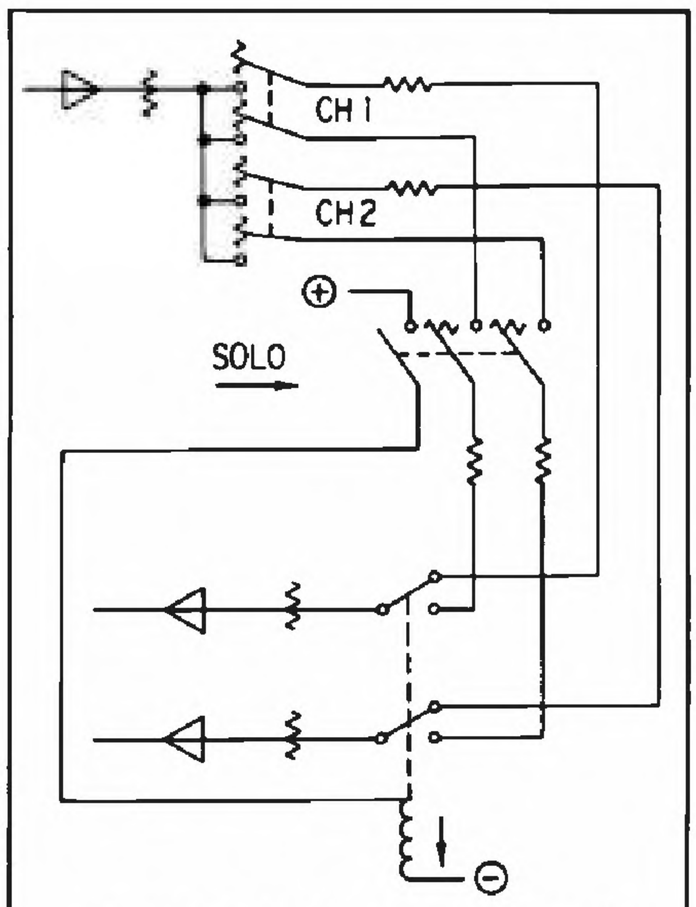

Figure 3. Selecting solo busses. In this single-line illustration, the switches are on terminating resistors.

combining networks are standard items, whether they be passive or active. As I prefer to have the phasing ability at this point, as well as at the input, the console uses a balanced buss of the "O" configuration. A separate key is also available which disables the regular busses, and—by relay—substitutes a set of busses called *solo*. (See FIGURE 3.) Because of the switching design, this solo position places the selected channel alone on the output, in its exact level and distribution among the output channels, as it would appear in the final mix. It is possible to achieve this effect without affecting the normal recorder outputs by moving the activating relays to the inputs of the control room monitoring system. This would then necessitate the addition of duplicate amplifiers and dual tracking faders in the output sections of the console.

The ability to hear a single performer or section is an added benfit to the mixer and producer during the pre-recording balancing. But whether this facility would be desired during the take, without disturbing the take, is again a matter of complexity and cost.

Under normal conditions the console's master channel faders are just that, but for special effects, or original mastering on monophonic/two channel, each of the four output channels has the added capability of being switched to any or all of the recorder busses. Thus the masters in effect become submasters. A final master could be added, to be effective only when the mixer desires the additional control. At present a board fade is not difficult as only four channels are present. Should expansion to eight occur, then the physical requirements would dictate a final master.

Finally, keys that are set for each recording machine permit deletion of the normal bussing, and substitution of the recorder into the master channels for playback and duplication/dub-down. The keys also disconnect the record feeds to that machine (to prevent a loop path, should the machine's monitors be left in *record* position). The *normal* play route for duplication and dub-down is through the high-level inputs on the input module. This routing permits the addition of additional equalization, echo, or other control conditioning that may be desired.

Ideas such as those above, and others incorporated in the console have been gleaned from exceptional consoles around the country, as well as designed for the special requirements of this application. It is desirable that other ideas that you may have devised, whether in this area or any other, be disseminated among others in the audio field. THE FEEDBACK LOOP is your sounding board for ideas, and your fact sheet for co-operation information.