Advanced Mixer Design.

This page last updated: 26 Apr 99. Diagrams improved.

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In 1990, Soundcraft introduced the Series 3200 recording console; probably still the best-performing mixer ever built. The electronic design work was done by myself and my colleague Gareth Conner. To celebrate what we felt to be a significant achievement, this article was written, and published in Wireless World (now Electronics World) in April 1991. All the electronic principles remain valid, and Soundcraft is still making the Series 3200. (as of March 1999) Things have moved on in the data storage area, and the references to tape machines now sound a little dated. Nonetheless, I have decided to leave it unchanged. It will be clear that this document merely scratches the surface of a subject that would fill a very large book, something like "The Whole Art of Mixer Design". Commercial considerations make it unlikely that such a book will ever be written.

Since the readership of WW covers a broad spectrum, familiarity with the recording process could not be assumed, and a concise account of it was published alongside the technical material below. This will be added to the site soon; watch for the "Multitrack Recording" department of The Institute.

- C O N T E N T S

MIXER DESIGN.

A large mixing console arguably represents the most demanding area of audio design. The steady advance of digital media demands that every part of the chain that takes music from performer to consumer must be near-perfect, as the comfortable certainty that everything will be squeezed through the quality bottleneck of either analogue tape or vinyl disc now looks very old-fashioned.

Competition to sell studio time becomes more cut-throat with every passing week, and it is clear that advances in console quality must not harm cost-effectiveness. The only way to reconcile these demands is to innovate and to keep a very clear view as to what is really necessary to meet a demanding specification; in other words the way forward is to use conventional parts in an

unconventional way, rather than simply reaching for the most expensive op-amp in the catalogue.

The technical problems that must be overcome in a professional mixing console are many. A large number of signals flow in a small space and they must be kept strictly apart until the operator chooses to mix them; crosstalk must be exceedingly low.

There will be up to 64 input channels, each with many stages, all have the potential to add distortion and noise to the precious signal. Even summing these signals together, while sounding trivially easy, is in practice a major challenge. In short, requirements are much more demanding than those for the most expensive hi-fi equipment, because degradation introduced at this stage can never be retrieved.

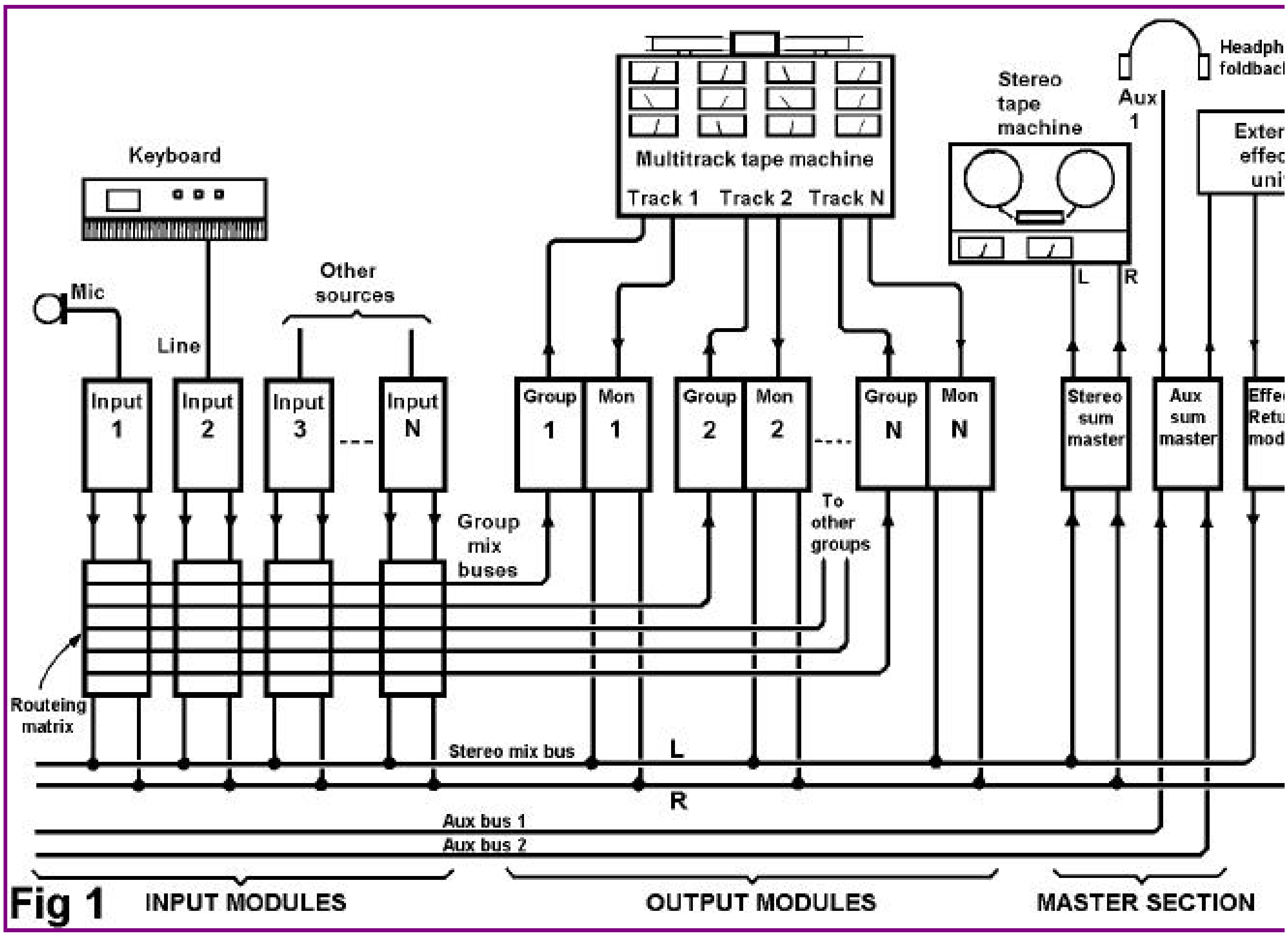


Fig l. System diagram of complete mixing console, showing division into inputs, group monitor contributions and master modules. Routeing matrix determines which group of inputs shall be fed to a given track on the multi-track tape machine. Several channels share one effects device.

The major functions of consoles are largely standardised, although there is much scope for detailed variation. Fig 1 shows the system diagram.

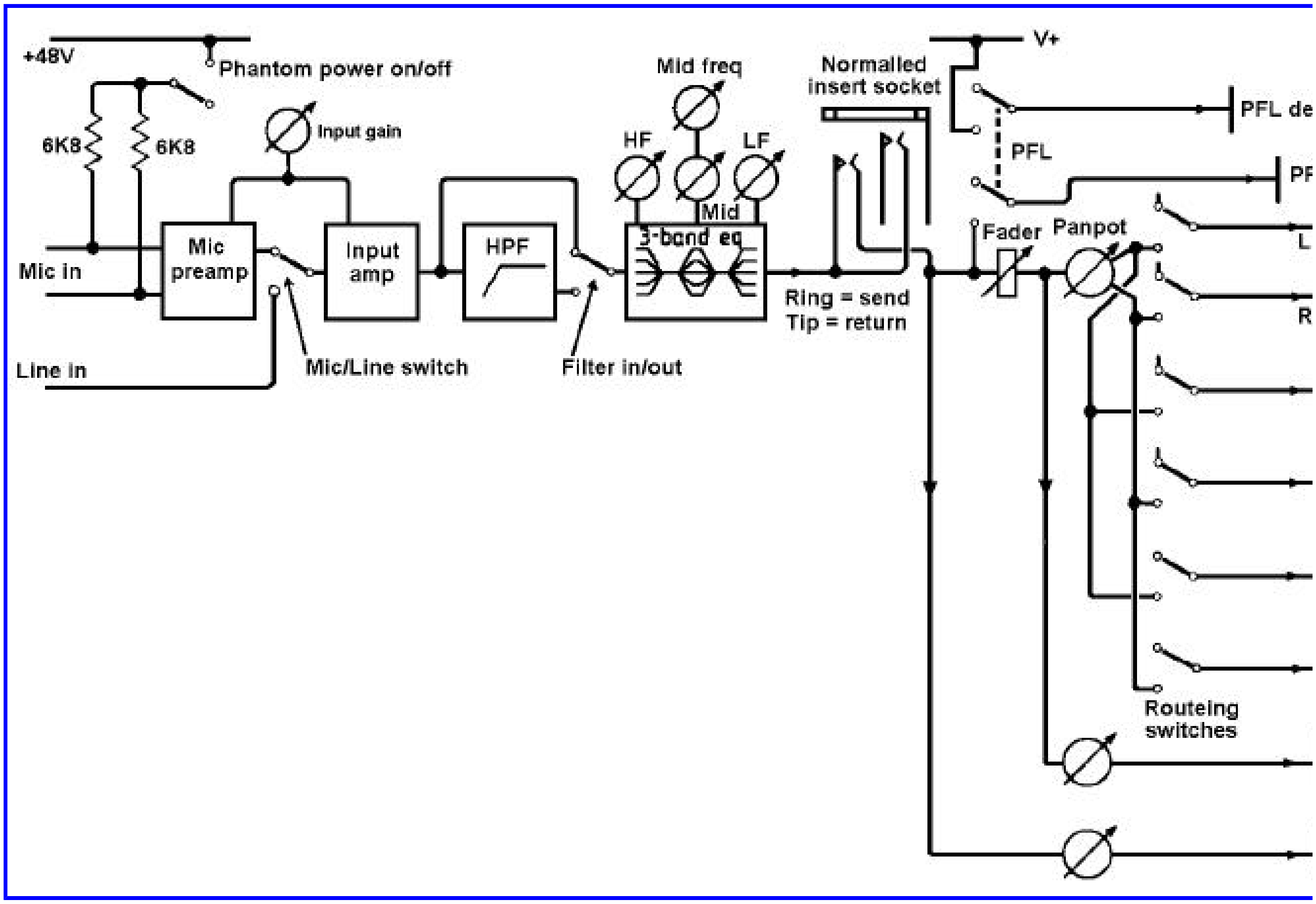
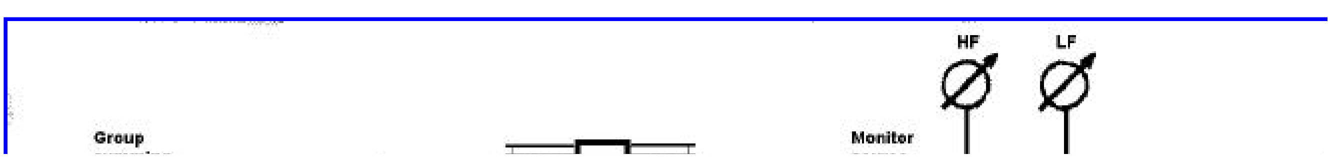


Fig 2. One input channel. Gain control is 70dB and tone control is standard Baxandall shelving type with addition of mid-range lift and cut. Two auxiliary sends are shown.

Fig 2 shows a typical input channel for a mixing console. The input stage provides switchable balanced mic and line inputs; the mic input has an impedance of 1K - 2K, which provides appropriate loading for a 200 Ohm mic capsule, while the line input has a bridging impedance of not less than 10K. This stage gives a wide range of gain control and is followed immediately by a switchable high-pass filter (usually -3dB at 100Hz) to allow removal of low-frequency disturbances.

The tone-control section (universally known in the audio business as "EQ" or EQualisation) typically includes one or more mid-band resonance controls as well as the usual shelving Baxandall-type high and low controls. Channel level is controlled by a linear fader and the panpot sets the stereo positioning, odd group numbers being treated as left, and even as right. The PFL (prefade-listen) switch routes the signal to the master module independently of all other controls; a logic bus signals the master module to switch the studio monitoring speakers from the normal stereo mix bus to the PFL bus, allowing any specific channel to be examined in isolation.



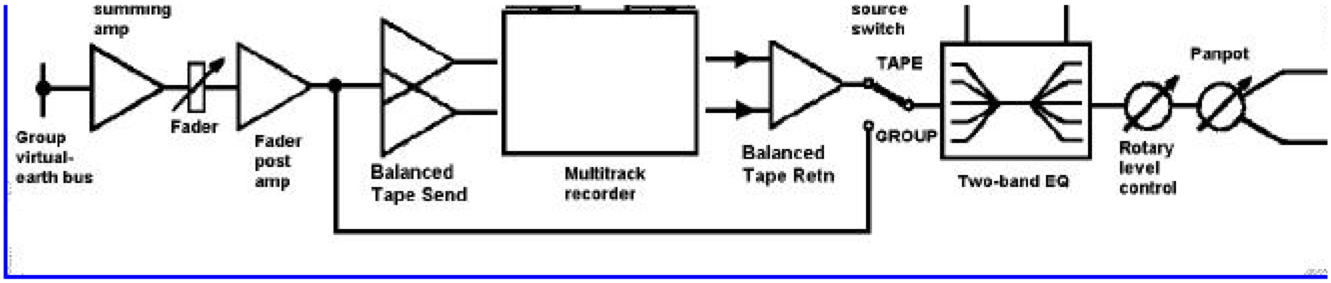


Fig 3. Block diagram of typical group module, showing switching between direct output and tape replay for monitoring purposes.

Fig 3 shows a typical group module and Fig 4 the basics of a master section; a manual source-select switch allows quality checking of the final stereo recording and two solid-state switches replace the stereo monitor signal with the PFL signal whenever a PFL switch anywhere on the console is pressed.

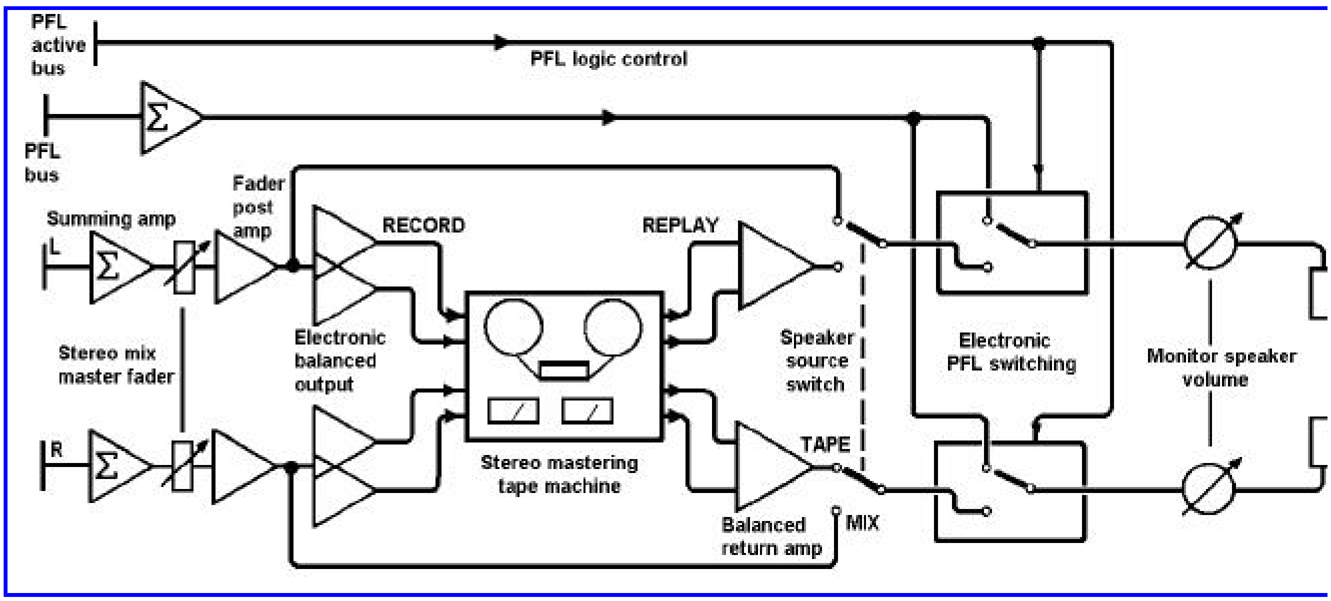


Fig 4. Block diagram of master module, with tape send/replay switching and electronic PFL switching.

AUXILIARY SENDS; FOLDBACK AND EFFECTS
The auxiliary sends of a console represent an extra mixing system that works independently of the main groups; the number and configuration of these sends have a large effect in determining the overall versatility of the console. Each send control provides a feed to an auxiliary bus; this is centrally summed and then sent out of the console.

Sends come essentially in two kinds: prefade sends, which are taken from before the main channel fader, and postfade sends, which take their feed from after the fader, so that the final level depends on the settings of both. There may be anything from one to twelve sends available, often switchable between pre and post. Traditionally, this means laboriously pressing a switch on every input module, since it is most unlikely that a mixture of pre and post sends on the same bus would be useful; the 3200 minimises the effort by setting pre/post selection for each bus from a master switch that controls solid-state pre/post switching in each module.

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Prefade sends are normally used for "foldback"; i.e. sending the artist a headphone feed of what he/she is perpetrating, which is important if electronic manipulation is part of the creative process, and essential if the artist is adding extra material that must be in time with that already recorded. In the latter case, the existing tracks are played back to the artist via the prefade sends on the monitor sections.

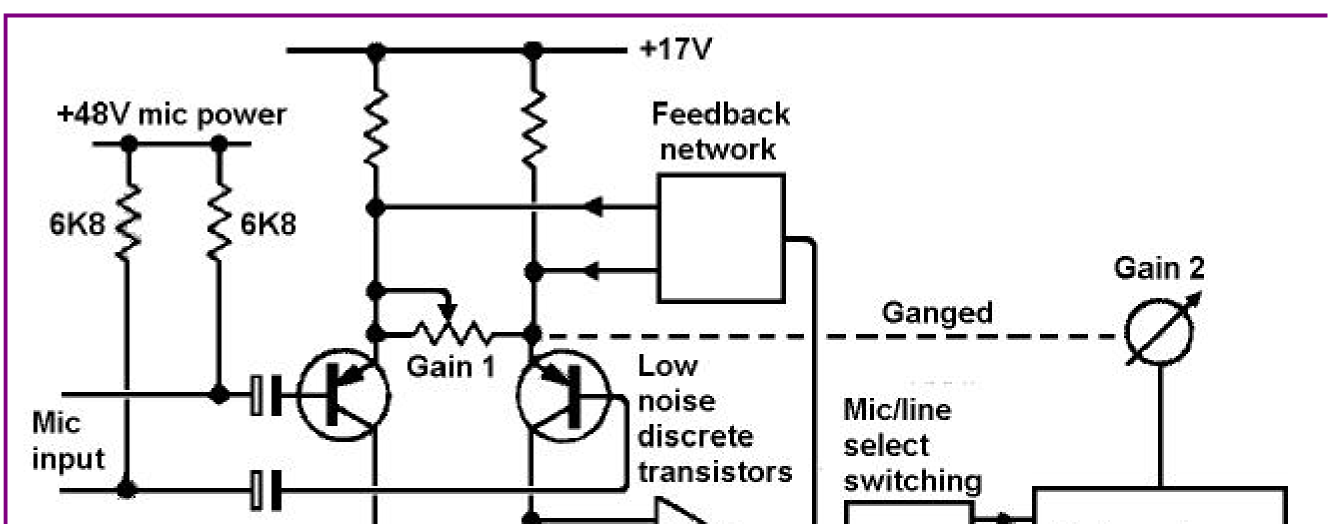
Postfade sends are used as effects sends; their source is after the fader, so that the effect will be faded down at the same rate as the untreated signal, maintaining the same ratio. The sum of all feeds to a given bus is sent to an external effects unit and the output of this returned to the console. This allows many channels to share one expensive device (this is particularly applicable to digital reverb) and is often more appropriate than the alternative of patching a processor into the channel insert point.

"Effect returns" may be either modules in their own right or a small subdivision of the master section. The returned effect, which may well now be in stereo, the output of a digital reverb., for example, is usually added to the stereo mix bus via level and pan controls. EQ is also sometimes provided.

MICROPHONE INPUTS.

The microphone preamplifier is a serious design challenge. It must provide from 0 to 70 dB of gain to amplify deafening drum-kits or discreet dulcimers, present an accurately balanced input to cancel noise pickup in long cables and generate minimal internal noise. It must also be able to withstand +48V DC suddenly applied to the inputs (for phantom-powering the internal preamps in capacitor mics) while handling microvolt signals. The Soundcraft approach is to use standard parts, which are proven and cost-effective through quantity production, in new configurations. The latest mic preamplifier design, as used on the 3200, is new enough to be covered by patent protection.

It is now rare to use input transformers to match the low-impedance (150-200 Ohm) microphone to the preamplifier, since the cost and weight penalty is serious, especially when linearity at low frequencies and high levels is important. The low-noise requirement rules out the direct use of op-amps, since their design involves compromises that make them at least 10 dB noisier than discrete transistors at low impedance.



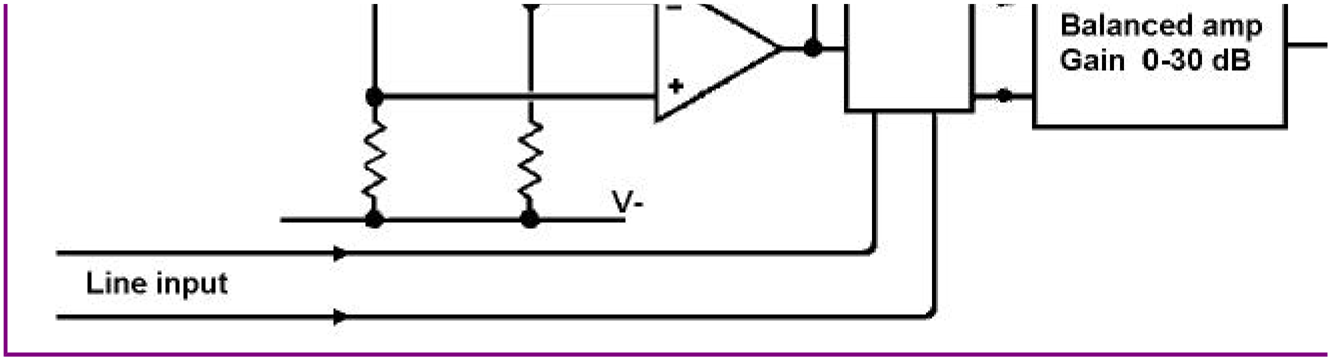
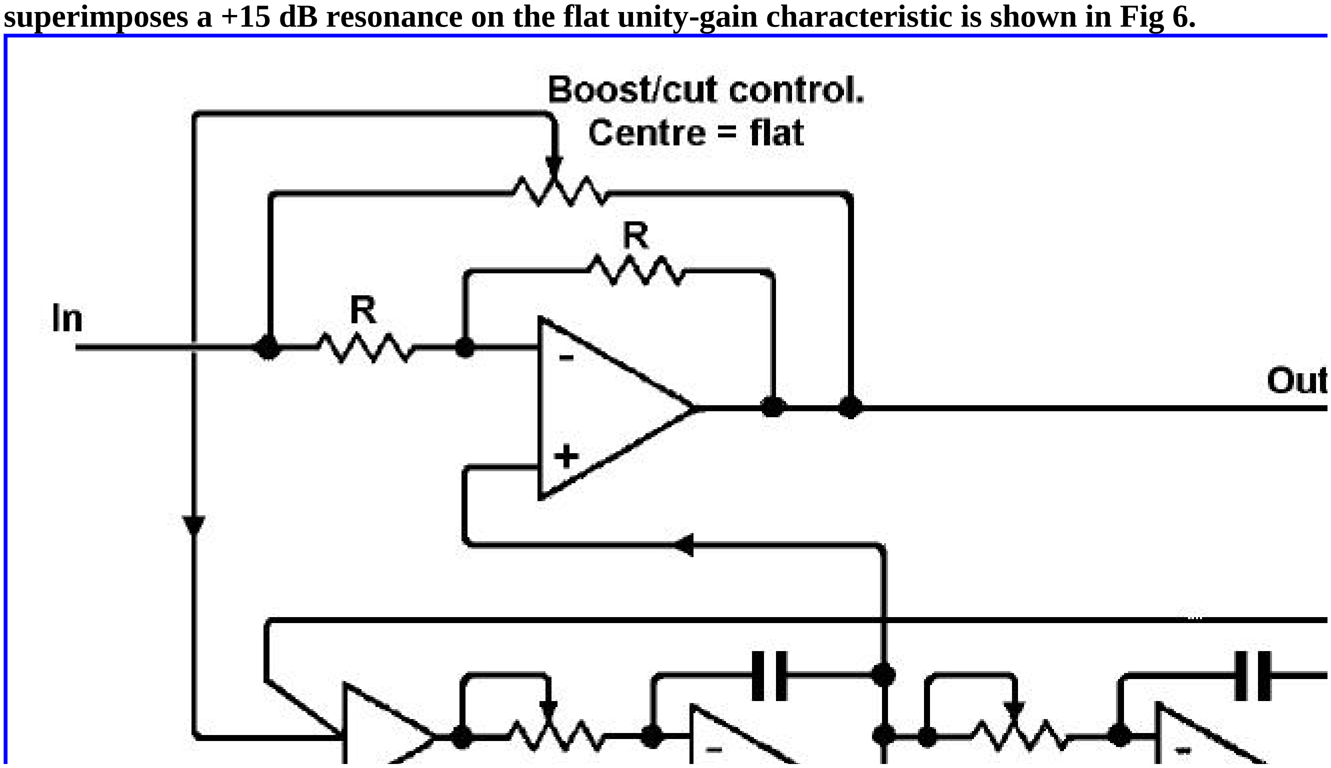


Fig 5. Low-noise microphone amplifier with wide gain range and balanced line output. Transistors in first stage avoid noise problem of op-amps.

This circuit, shown in Fig 5, therefore uses a balanced pair of low-noise, low-Rb PNP transistors as an input stage, working with two op-amps to provide load-driving capability and raw open-loop gain to linearise signal handling. Preamplifier gain is spread over two stages to give a smooth 0-70dB gain range with the rotation of a single knob. This eliminates the switched 20dB attenuator that is normally required to give the lower gain values, not only saving cost and complication, but also avoiding the noise deterioration and CMRR degradation that switched attenuators impose. The result is an effective input stage that is not only quieter, but also more economical than one using specialised low-noise op-amps.

THE EQ STAGES.

Since large recording consoles need sophisticated and complex tone-control systems, unavoidably using large numbers of op-amps, there is a danger that the number of active elements required may degrade the noise performance. A typical mid-band EQ that superimposes a +15 dB resonance on the flat unity-gain characteristic is shown in Fig 6.



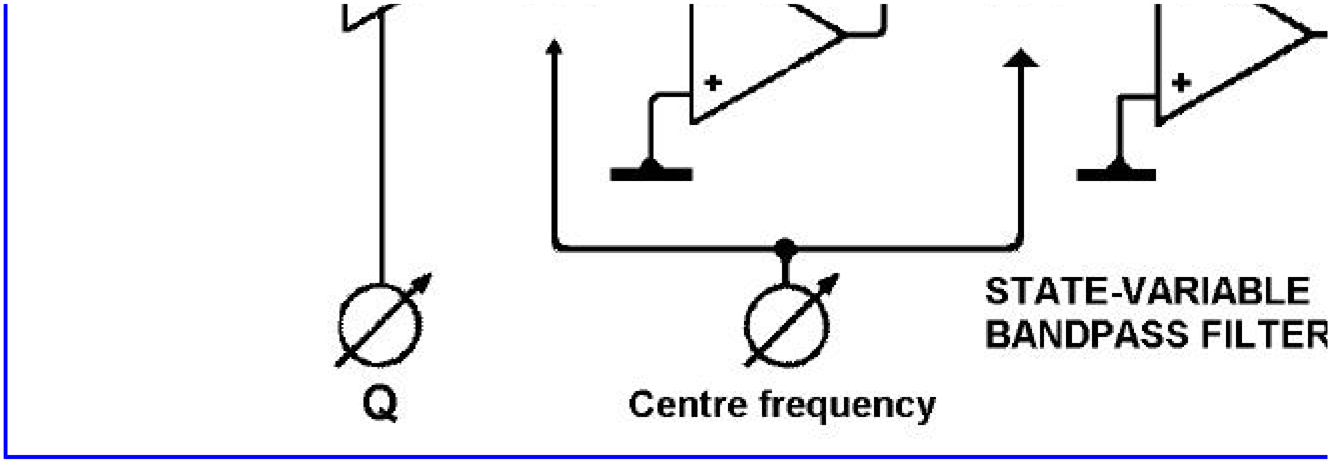


Fig 6. Parametric mid-band EQ stage. EQ and centre frequencies are independently variable, being set by the parameters of the state-variable filters.

A signal is tapped from the forward path, put through a state- variable band-pass filter which allows control of centre-frequency and Q, and then added back. To improve noise performance, the signal level at all locations (in all conditions of frequency, Q, and boost/cut) was assessed, and it proved possible to double the signal level in the filter over the usual arrangement, while maintaining full headroom. The noise generated is thus reduced about 6dB.

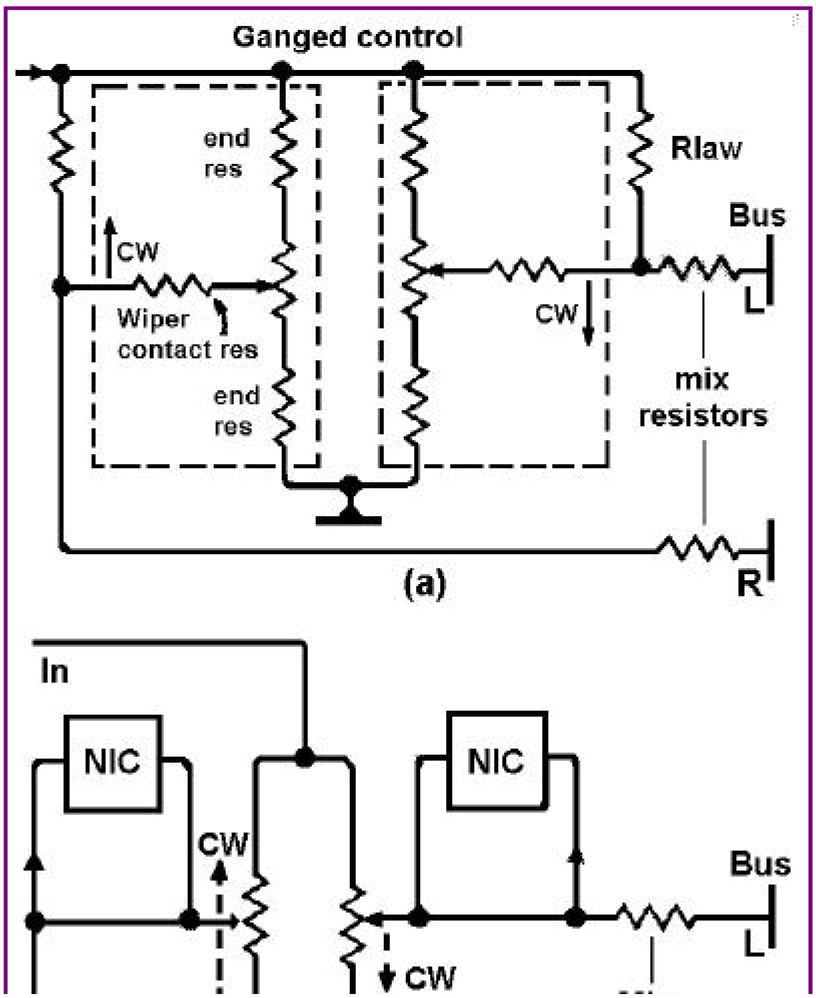
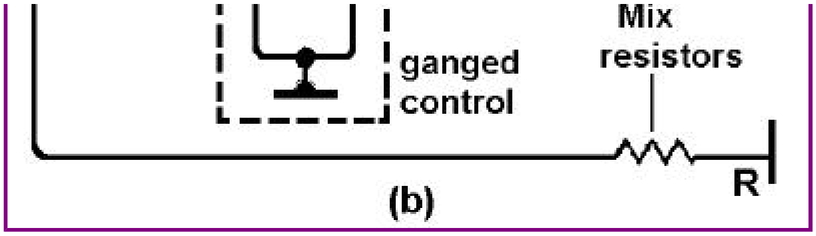


Fig 7. Standard panpot circuit at (a) showing how pull-up resistor Rlaw draws current through wiper contact resistance, which is usually greater than the end resistance of the pot, limiting maximum attenuation. Arrangement at (b) uses NICs to replace pull-up to modulate law with panpot setting. Left/right isolation increased from -65dB to -90dB.

THE PANPOT

To give smooth stereo panning without unwanted level changes, the panpot should theoretically have a sine/cosine characteristic; such components exist, but they are prohibitively expensive and so most mixing consoles use a dual linear pot. with its law bent by a pull-up resistor, as shown in Fig 7a. This not only gives a mediocre approximation to the required law, but also limits the panning range, since the pull-up signal passes

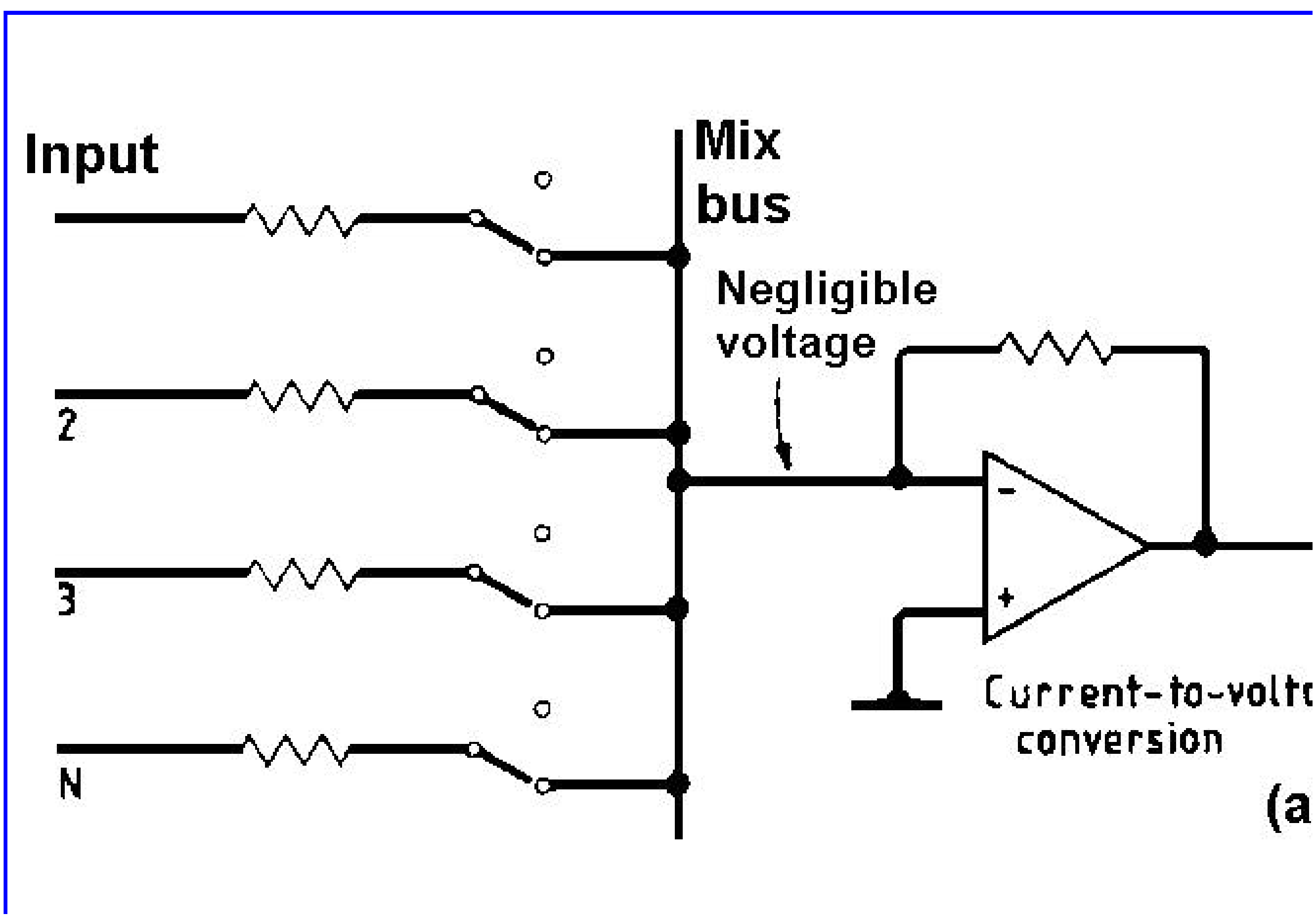


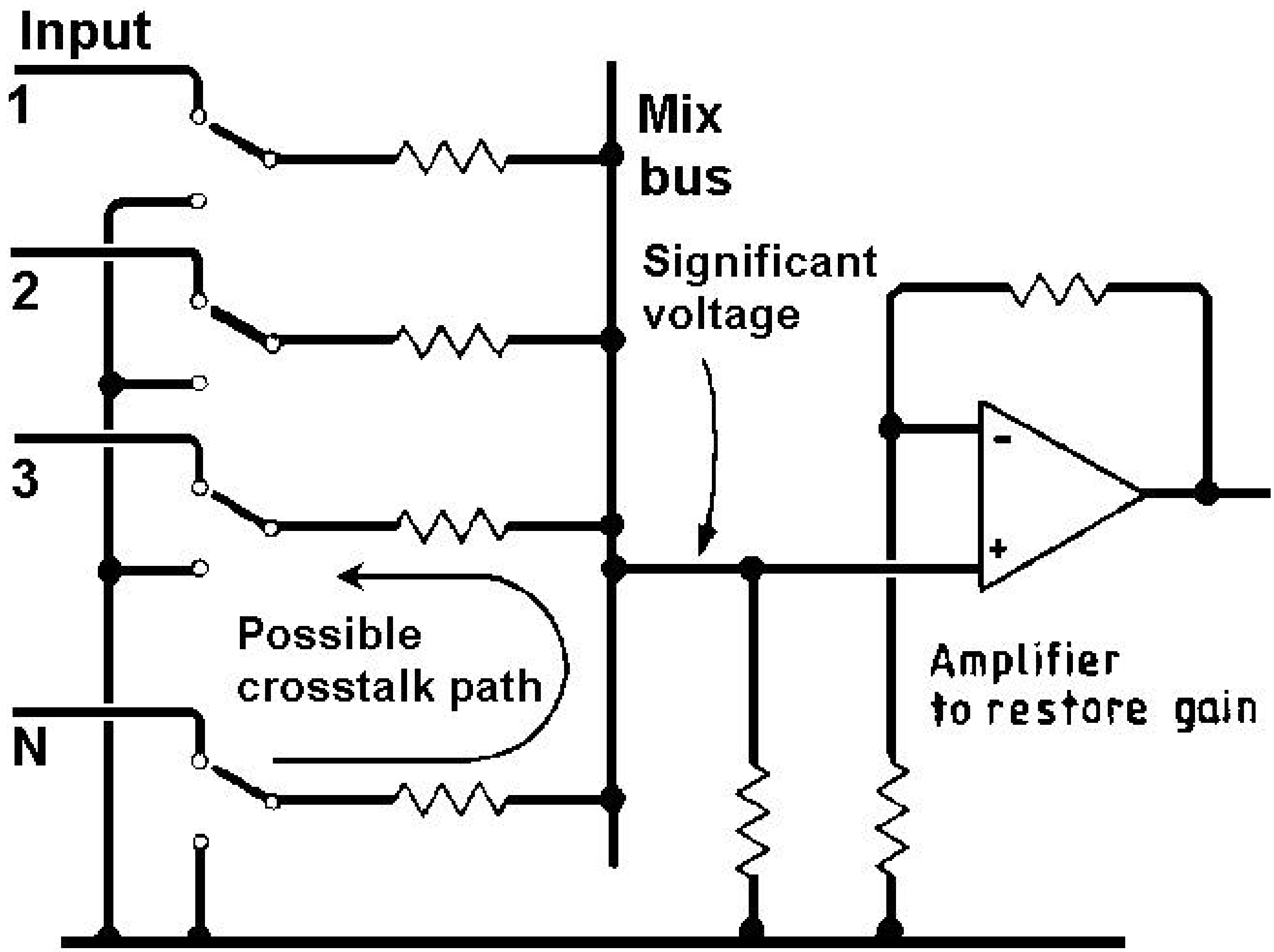
through the wiper contact resistance (usually greater than the end-of-track resistance) and limits the attenuation the panpot can provide when set hard left or right.

This limitation is removed in the Soundcraft active panpot shown in Fig 7b by replacing the pull-up with a negative-impedance-converter that modulates the law-bending effect in accordance with the panpot setting, making a close approach to the sine law possible. There is no pull-up at the lower end of the wiper travel, when it is not required, so the left-right isolation using a good-quality pot. is improved from approx -65dB to -90dB. This concept has also been made the subject of patent protection.

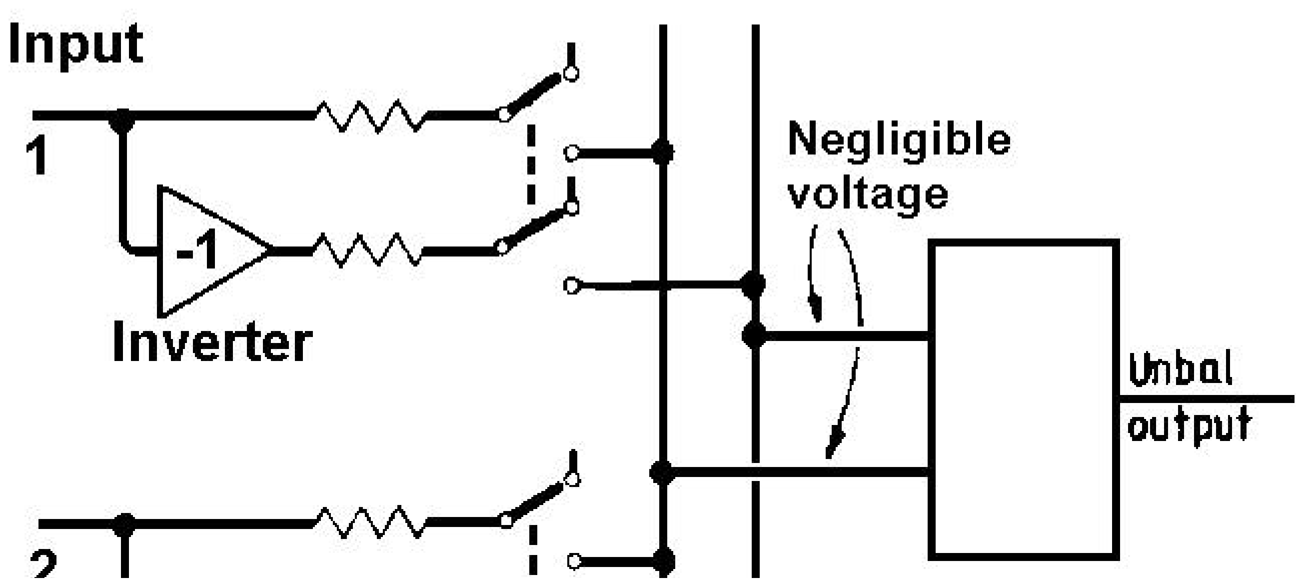
SUMMING TECHNOLOGY.

One of the biggest technical challenges in console design is the actual mixing of signals. This is done almost (but not quite) universally by virtual-earth techniques, as in Fig 8a. A summing amplifier with shunt feedback is used to hold a long mixing bus at apparent ground, generating a sort of audio black hole; signals fed into this via mixing resistors apparently vanish, only to reappear at the output of the summing amplifier, as they have been summed in the form of current. The elegance of virtual-earth mixing, as opposed to the voltage-mode summing technique in Fig 8b, is that signals cannot be fed back out of the bus to unwanted places, as it is effectively grounded, and this can save massive numbers of buffer amplifiers in the inputs.




Input
1
2
3
N
Mix
bus
Significant
voltage
Possible
crosstalk path
Amplifier
to restore gain

(b)


Input
1
-1
Inverter
2
Negligible
voltage
Unbal
output

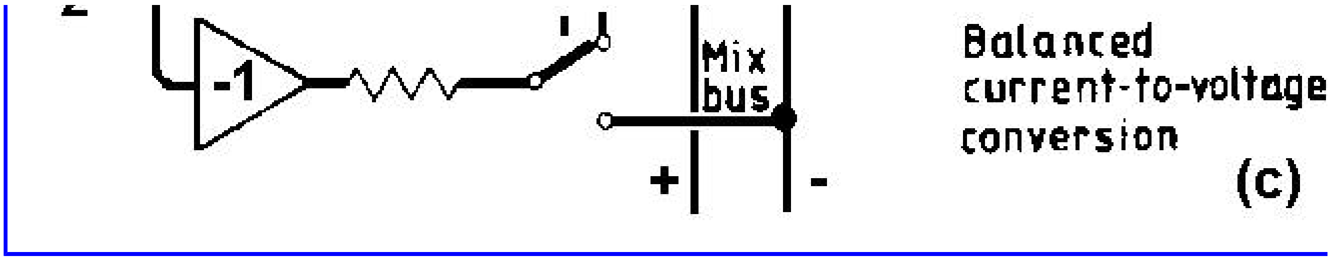


Fig 8. Virtual-earth summer at (a) effectively eliminates cross-talk, since there is almost no signal at the summing point. Voltage-mode circuit at (b) allows cross- talk. Balanced virtual-earth summing circuit at (c) requires a separate inverter for each channel to provide the anti- phase signal.

There is, however, danger in assuming that a virtual earth is perfect; a typical op- amp summer loses open-loop gain as frequency increases, making the inverting input null less effective. The 'bus residual' (i.e. the voltage measurable on the summing bus) therefore increases with frequency and can cause inter-bus crosstalk in the classic situation with adjacent buses running down an IDC cable.

Increasing the number of modules feeding the mix bus increases the noise gain; in other words the factor by which the noise of the summing amplifier is multiplied. In a large console, which might have 64 inputs, this can become distinctly problematic. The Soundcraft solution is to again exploit the low noise of discrete transistors coupled to fast op-amps, in configurations similar to the mic preamps. These sum amplifiers have a balanced architecture that inherently rejects supply- rail disturbances, which can otherwise affect LF crosstalk performance.

As a console grows larger, the mix bus system becomes more extensive, and therefore more liable to pick up internal capacitive crosstalk or external AC fields. The 3200 avoids internal crosstalk by the use of a proprietary routeing matrix construction which keeps the unwanted signal on a bus down to a barely measurable 120dB. This is largely a matter of keeping signal voltages away from the sensitive virtual-earth buses. Further improvement is provided by the use of a relatively low value of summing resistor; this also keeps the noise down, although since it drops as the square-root of the resistor value, at best, there is a clear limit to how far this approach will work before drive power becomes excessive; 4.7K is a reasonable minimum value.

External magnetic fields, which are poorly screened by the average piece of sheet steel, are rejected by the balanced nature of the 3200 mix buses, shown in Fig 8c. The operation is much the same as a balanced input; each group has two buses, which run physically as close together as possible and the group reads the difference between the two, effectively rejecting unwanted pickup. The two buses are fed in antiphase from each input, effectively doubling the signal level possible for a given supply voltage. Overall mixing noise is reduced by 3dB, the signal level is 6dB up and the noise, being uncorrelated for each bus, only increases by 3dB.

The obvious method of implementing this is to use two summing amplifiers and then subtract the result. In the 3200, this approach is simplified by using one symmetrical summing amplifier to accept the two antiphase mix buses simultaneously; this reduces the noise level as well as minimising parts cost and power consumption. The configuration is very similar to that of the balanced mic amp., and therefore gives low noise as well as excellent symmetry.

SOLID-STATE AUDIO SWITCHING.

There are two main applications for electronic switching in console design. The first is "hard" switching to reconfigure signal paths, essentially replacing relays with either JFETs (Fig 9a) or 4016-type CMOS analogue gates which, since they are limited to 18V rails and cannot handle the full voltage swing of an op-amp audio path, must be used in current-mode, as shown in Fig 9b. Note that when gate 1 is off, gate 2 must be on to ensure that a large voltage does not appear on gate 1 input. Full-voltage range analogue gates do exist but are very expensive.

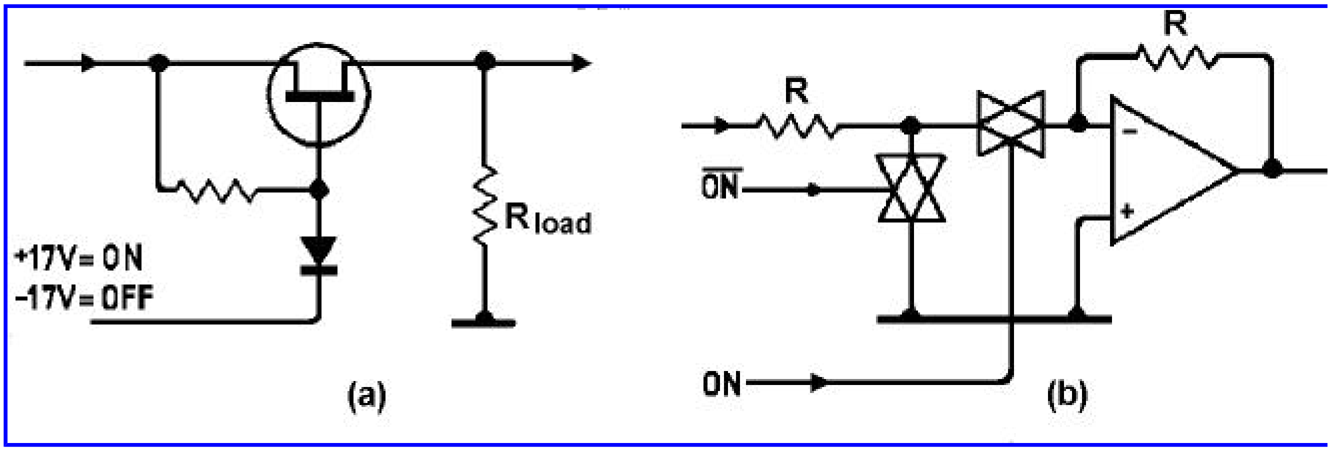


Fig 9. Hard switching with JFETs in voltage mode (a) and with analogue gates in the current mode (b), which prevents gate elements from being driven outside their voltage capabilities.

Secondly, there is channel muting; this not a hard switch, since an unacceptable click would be generated unless the signal happened to be at a zero-crossing at the instant of switching; the odds are against you. The 3200 therefore implements muting as a fast-fade over 10ms; this softens transients into silence while preserving time-precision. It is implemented by a series-shunt JFET circuit, with carefully timed and synchronised ramp voltages applied to the FET gates.

PERFORMANCE FACTORS.

Primary requirements of modern consoles are very low noise and minimal distortion. Since a comprehensive console must pass the audio through a large number of circuit stages (perhaps over 100 from microphone to final mixdown) great attention to detail is essential at each stage to prevent a build-up of noise and distortion; often the most important trade-off is the impedance of the circuitry surrounding the op-amp, for if this too high Johnson noise will be increased, while if it is too low an op-amp will show degraded linearity in struggling to drive it.

The choice of device is also critical, for cost considerations discourage the global use of expensive chips. In a comprehensive console like the 3200 with many stages of signal processing, this becomes a major concern; nonetheless, after suitable optimisation, the right-through THD remains below 0.004% at 20dB above the normal operating level. At normal level it is unmeasurable.

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