Noise & Headroom in Mixers.

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NOMINAL SIGNAL LEVELS.
Signals vary continuously in their actual amplitude, so it is necessary to specify a nominal or "normal" signal level. Explicitly or otherwise, this assumes a continuous sine wave of a specified RMS voltage.
The level may be quoted in Volts RMS but is more commonly expressed in decibels, eg dBu or dBv. Decibels alone are merely a ratio, so they must be based on a reference if they are to refer to an absolute rather than a relative level.

As an example, professional mixing consoles use a standard nominal level of +4 dBu, equivalent to 1.228 V. The "u" means that the reference level is 775 mV. This rather odd-looking value was chosen because it gives 1 mW of power in a 600 Ohm load. (The 600 Ohm load is a historical hangover from line transmission systems, as it is the characteristic impedance of spaced copper wires on a standard-sized telegraph pole; you don't need to worry about this)
This could also be expressed as +1.8 dBv, the "v" meaning a reference level of 1 Volt.

The nominal level is sometimes the maximum that can be recorded on a given medium, but this is not true for electronic systems such as mixing consoles, where the maximum possible level is often 20 dB or more above the nominal level.

NOISE.
Noise is the hiss in the background. All electronic equipment adds a (hopefully) small amount of noise to the signals being handled. This is unavoidable, and the measure of a quiet piece of equipment is that it adds as little noise as possible.

The word "noise" is sometimes used to include any hums, buzzes or other extraneous sounds the system may be subject to; this is not very helpful as it confuses inevitable white noise with interference that is avoidable with correct design.

The noise level is often called the "Noise Floor" as it sets the lower limit to the dynamic range. This does not mean that anything lower in level than the noise floor is totally inaudible; for example a hard-edged buzz would probably audible (given enough amplification) even if it was 20 dB below the noise floor.

Noise is generated in every conductor or component that has resistance. In addition to this, all semiconductors or other active components add extra noise of their own. From this it may sound as though it is very hard to make a quiet piece of equipment; but in practice most of these noise sources make a negligible contribution, and only a few need close attention.

This is because the various source of noise are all uncorrelated- ie they are all random, but differently random, and so when the are added together some cancellation as well as summation occurs. If two correlated signals of equal amplitude are added together, the total amplitude is doubled. (6 dB increase) In contrast, if two different sources of white noise are added, the level only goes up by 3 dB. This is mathematically known as RMS summing.
If one of the noise sources is reduced in amplitude, its contribution to the total drops away very quickly:

Source 1	Source 2	Summed total
0 dB	0 dB	+3.01 dB
0 dB	-1 dB	+2.54 dB
0 dB	-2 dB	+2.12 dB
0 dB	-3 dB	+1.76 dB
0 dB	-6 dB	+0.97 dB

Roughly speaking, one dB is the smallest perceptible change in sound level. Therefore adding any noise source that is less than half the noise level existing already makes very little difference.

HEADROOM.

Every electronic channel, be it digital or analogue, has a maximum signal level it can pass. Attempts to exceed this typically result in hard clipping, where the waveform is levelled off flat when it reaches the limits.
The maximum level is much harder to define for analogue tape machines. Here a higher recording level can be obtained by accepting some distortion on signal peaks. This is possible because it is low-order distortion, generating mostly third harmonics. Hard clipping such as occurs in electronic circuitry or digital storage, generates lots of high-order harmonics that are far more obtrusive. Hard clipping is not normally detectable if it is very brief (say 1 millisecond) but if sustained sounds very unpleasant.
Both in recording and live music, the level of signals arriving is not entirely predictable. It is clearly desirable to have some safety margin between the "usual" operating level and the clipping point.

DYNAMIC RANGE.

The difference between the maximum level and the noise floor is known as the dynamic range. In electronic circuitry this is usually 100 dB or more, the exception being mic amplifiers working at high gain.

	Noise floor	Headroom	Dynamic range
Analogue tape		0 dB	
Digital	-83 dBu	+10 dB	93 dB

Mixing console	**-84 dBu**	**+26 dBu**	**110 dB**

THE ASSESSMENT OF NOISE IN MIXING CONSOLES.

The noise performance of a complex system such as a mixing console is not simply described. An almost infinite number of variations in console set-up, and measurement technique and interpretation exist. The final noise level at an output depends on the operational configuration in use as much as on the technological decisions made at the design stage, and so this document has been produced to clarify some of the measurement philosophies and techniques used in the design, testing, and specification of consoles.

Most of the actual data quoted are taken from the Soundcraft Series 6000, as one of the quietest consoles ever built, but the general principles are applicable to all mixing consoles. It is assumed that the reader is familiar with audio basics such the difference between dBu and dBv.

In recording work the most critical situation is mixdown, as the maximum number of potential noise sources are being added together. The situation is simplified because all inputs are line inputs, and these will be set at or near unity gain, assuming the levels of console and tape machine are properly aligned. Where inputs are being mixed in directly from sequenced synthesisers, these are likely to be noisier than any line input amplifier, and the discreet use of noise gating is not unusual.

Consoles for PA work are normally used in a way that approximates to mixdown in a recording console, as a large number of inputs are mixed down to stereo. However in this case many of the inputs will be used for microphones, probably with widely varying input gains, and the situation becomes more complex. In general it seems impossible to specify one particular configuration that would provide a realistic assessment of noise performance in all live situations.

NOISE AND THE LAYING OF TRACKS.

Soundcraft noise specifications are set using true-RMS readings. It is not our practice to quote noise measurements made with weighting filters (such as A-weighting) because it is our feeling that this would tend to mislead people. Once again, it would also limit the choice of test gear that can be used to check that a console is performing to specification.

The last two points apply to all our noise measurements, and not just those involving input preamps.

We shall now examine the noise structure of a Series 6000 console in recording mode. Typically the mic input gain might be set to +50dB, and at this setting the preamp EIN is still as good as -127 dBu. From this we can quickly find that that the preamp noise output is -77 dBu, and while this may seem high at first sight, even an absolutely noiseless circuit could only improve this by 2 dB. The EQ section that follows the preamp generates less than -95 dBu, and therefore its contribution is completely negligible.

Similarly the noise contribution from the fader post-amp is approximately -104 dBu, and this makes no measurable

**difference.
Assuming that one channel is routed to a group in order to send it to the multitrack, the extra noise from the group summing amplifier will be about -101.5 dBu, which is once more a negligible increase.**

It can therefore be clearly seen that in recording mode, the performance of the Series 6000 is determined to a very large extent by noise from outside the console, in the shape of the inevitable microphone Johnson noise.

NOISE AT MIXDOWN AND IN PA WORK.

At mixdown there are typically many inputs, all feeding the stereo mix bus in varying amounts. The most obvious of these are the input channels, as considered above, but it is important not to forget the presence of the monitor sections, which may well have a different noise characteristics. The same applies to any effect returns routed to the stereo bus. Each input will have varying frequency characteristics when the EQ is in use, and if input are sub-grouped this adds an extra layer of complication.

The number of variables is large, and the interactions between them complex, and so we have attempted to standardise a measurement configuration that would, as closely as possible, represent an 'average' mixdown insofar as such an animal exists. The basis of this configuration is the recognition that if 16 or 24 input channels are all mixed into the stereo bus with their faders at 0dB, the result is likely to be an inharmonious crunching as the mix summing amplifier clips. For most of our tests we have chosen to use 16 inputs as this is possible on more types of console, though the principle is of course the same for 24 or more.

Assuming that the inputs are uncorrelated, their voltages will sum in an RMS fashion- in other words the result is not the arithmetic sum but the square-root of the sum of the squares. Therefore 16 inputs will combine to give a signal roughly 4 times greater (the square root of 16) and somewhere in the system attenuation must be introduced to reduce the signal to the nominal level; in other words, the inputs are summing to unity. We felt that it was more likely that this should be introduced by pulling down the channel faders to -12 dB, (rather than pulling down the master fader) as this protects the mix summing amps from overload.

In fact, mix summing overload is very unlikely on the Series 6000, as the discrete/integrated summing-amp technology used allows the amplifiers to be run at a low gain without sacrificing noise performance. This gain structure also conveniently allows the mix inserts to run at a nominal -10 dBv. (equal to -7.8 dBu, sometimes called Tascam level)

Other decisions had to be taken to define the test configuration. Line gain is set to unity, ie +4 dBu in for nominal internal level, and the line inputs are terminated in a short circuit as the output impedance of professional equipment feeding it is unlikely to be higher than 100 Ohms. This is negligible compared with the line amps internal impedances. (It is not inappropriate here to underline the fact that the output impedance of an amplifier does not determine its current-drive capability- it is possible to have

a stage with a 10 Ohm source impedance that cannot fully drive a load below 10 kOhm)

Since the relative amounts of cut and boost applied are impossible to predict, it was decided that EQ should be switched in but set flat. The control centre-detents help here as the configuration must be relatively quick to set up, and repeatable.

Our normal procedure is to begin measurements at the mix outputs, with the mix fader down. This parameter is important because it determines the ultimate quietness reached when doing an overall fade-out. Then the audio path is successively extended to the 16 inputs, first by fully advancing the mix fader to 0 dB, then routing the channels to mix with channel faders down, and then advancing these faders to the previously described setting of -12 dB. Finally the EQ section is switched in. This stepwise process gives a very clear picture of how noise builds up through a console. Note that the monitor sections are switched on, and panned hard left so that their effect can be determined by recording the results for both left and right mix outputs. Likewise the channels are panned hard left to discriminate between mixing noise and noise from the fader post-amp. The figures below are actual measurements from a Series 6000 prototype.

	LEFT	RIGHT
Mix fader down.	-102 dBu	-102 dBu
Mix fader 0 dB.	-85.7 dBu	-87.5 dBu
16 channels routed.	-83.5 dBu	-84.5 dBu
Chan faders to -12 dB.	-81.5 dBu	-84.5 dBu
EQ section in.	-81.0 dBu	-84.5 dBu

All channels and monitor sections are panned fully Left.

DIGITAL EQUIPMENT AND NOISE.

When digital recording systems were first introduced they were nominally 16-bit. This gives a theoretical noise floor 96 dB below maximum level. In practice 90 dB was more likely, and if maximum level corresponds to +4 dBu, this indicates a noise level at each multitrack output of -86 dBu.

It is necessary now to consider the amount of headroom left when the nominal operating levels are set up. Unlike analogue tape, digital clips suddenly and with painful clarity, and so it is essential to leave enough headroom to prevent this. If 10 dB of headroom is left on all tracks that are not completely predictable in level, then this implies that the noise level actually present at the console line inputs is -76 dBu. After this level has been passed through channel faders set to -12 dB as above, and then summed to unity, it remains at -76 dBu, well above any of the noise readings quoted in the above table. Thus the console will be at least 5 dB quieter than the digital equipment it is connected to, even in the worst case. It is of course more

likely that the panpots are on average central, which yields a larger margin of about 8 dB.

Many digital systems now have 20-bit or more resolution; though in general only 20 bits is achieved in practice. This reduces the theoretical noise floor by 4 bits, or 24 dB. Therefore the noise level at the recorder output should be -110 dB. In practice the analogue input/output circuitry of the recorder is unlikely to be able to achieve this, and assessment of the relative noise performances of recorder and console must be done on a case-by-case basis.

Douglas Self

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