Noise in power supplies – Part 2

As the long-awaited follow up to his definitive study of switch-mode power supply noise in Resolution V8.2, TONY WALDRON takes his findings further and has worked out a way to measure it.

In a previous episode... daring and courageous sound designer Mike Walker telephoned ancient (but intrepid) mixing console designer Anton Waldron, telling him that the noise in switch-mode power supply units could be detected in an A/B test with an old linear power supply unit. Further listening tests revealed that this was indeed the case. Ancient (but still intrepid) Anton confirmed that switch-mode power units coupled high levels of high frequency noise into the DC power lines of the audio circuits in the mixing console. However — at the time — Anton was unable to prove that switch-mode power supply noise was in any way present in the output of the audio circuits...

Since my first attempt at detecting the result of switch-mode power supply noise in the output signal of an audio circuit (Resolution V8.2), I have made a number of improvements in the design of our production switch-mode power units. Thus, I didn’t want to use a latest design model in the second series of tests right away. As luck would have it, an early design switch-mode unit was returned to the factory for repair, so I seconded it for service as soon as it was available. An equivalent sized linear power unit was also to hand, so my second series of tests could begin.

The switch-mode unit used for the Part 2 tests was a 2008-built 8500 power supply, as used with the CADAC S-Type mixing console. This power supply is based on a reasonably well behaved switching block made by Lambda (now TDK Lambda), producing ±18V at 15A, +13V at 12A, and 48V at 1A.

The linear unit was a power supply designed for the old CADAC A-Type console and was built around 1979. This PSU was able to produce ±18V at 12A, ±13V at 10A, and 48V at 1A.

The switch-mode and the linear power units were connected simultaneously to an S-Type mixing console using the main and backup power input ports (switch-mode connected to main, linear connected to backup). This arrangement (Figure 1) allows the audio circuits in the mixer to remain powered up all the time. The quality of the audio output signals can be compared by simply turning off one of the power units. The S-Type console was a 53-slot version, fitted 24 input channels, 8 groups, and one listen module. The current drawn on each power rail by the console was units. The S-Type mixing console was a 33-slot version, fitted 24 input channels, 8 groups, of the audio output signals can be compared by simply turning off one of the power rails.

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This is well within the capability of either power supply unit and this is important, because power supply output noise increases with the output current.

Table 1 shows the residual noise on the power supply output rails used by the audio circuits. These results were obtained on a power supply load test rig that monitors output noise on an oscilloscope and an old analogue precision Volt meter, as well as the amount of current being supplied to the load. It was found that the switch-mode power unit produced a lot more residual noise than the linear design — just as was found in Part 1 (Resolution V8.2). In this case, the switch-mode noise was some 12dB (more than three times) greater than the linear on the +18V rail. But the noise on the -18V rail was 24.8dB (more than 17 times) greater! Similarly, the colour of the residual noise was quite different between the two power units. I always listen to the audio content when I do noise tests. The sound on the linear power output rails was clear ‘hiss’ (white noise), but the switch-mode output lines had a lot of RF twittering superimposed on the hiss.

Even though the RMS noise levels are very low with respect to the power supply rail voltages (see Table 2), my experience is that people working with sound every day are easily able to resolve very small differences during A/B tests. When the noise quality changes at the same time, I have no doubt that even smaller differences can be resolved.

To measure differences in the audio output when the power units were changed over, I chose the Audio Precision AP2 analyser that I also use during circuit and system development. This instrument has very low self noise and distortion, and can do all of the normal analogue test procedures. It can also perform Fast Fourier Transform (FFT) analysis.

ANALYSIS 1 — Using the test setup as shown in Figure 1, my first job was to check that all of the electronics involved in the tests was working normally. I quickly established that the Audio Precision AP2 and the mixing console were performing to their specifications. Indeed, the mixing console gave consistent results for frequency response and output noise (band limited 22Hz to 22kHz), no matter which power unit was in use at the time.

The first measured difference of any significance occurred during the distortion tests. When the mixing console was powered by the switch-mode unit, the distortion was always 5 times greater than when powered by the linear. I did the same distortion test with every input channel routed to all eight output groups in turn.

The difference in distortion was always the same — some 5 times greater when the switch-mode power unit was driving the electronics than with the linear. A typical result is shown in Figure 2. Here, the audio output distortion when powered by the linear unit is the red curve. You can easily see that the difference in distortion is unequivocal. Also note that I do most audio tests with an increased bandwidth — 10Hz to 100kHz — just in case there are resonance problems with the circuits or system setup.

Of course, even when the mixer is powered by the switch-mode unit the distortion is very low. I am sure that some will look at the figures and feel that it is impossible for anyone to detect such a small difference even though one small number is 5 times greater than the other small number. To test this out, I set up another A/B listening test with the same power unit setup. The listening panel included two musicians, both with recording and live sound mixing experience, and two design engineers — the last two represented listeners with normal hearing.

Now it is well known that the power of suggestion is very strong indeed, which explains why people can ‘hear’ real benefits when large glossy and expensive cables are connected to their speakers, compared with bell-wire (adjusted for level). So, for the new listening tests, the monitor loudspeakers and listeners were assembled in a double-blind test concept. Communications between the rooms was conducted by our student opening the adjoining door and shouting results to me.

Knowing that it is essential to establish exactly the same levels for device A and device B in any A/B test, I spent some time in accurately calibrating the output levels fed to the monitor system before the start of the test. During the test, I played line level source material without the mixing console in circuit; the same source material via the mixing console powered by the switch-mode power unit; and the same source material via the mixing console powered by the linear power unit. The order of the three configurations was varied on each playback sequence.

Complaints that the stereo image had deteriorated occurred each time the source material came via the combination of console and switch-mode power. There were no comments about any changes in the source image between console with linear power or source direct to monitor system.

AP2 sweep analysis of each channel was conducted at the end of each set of the three configurations. This confirmed the difference in distortion already noted. Repetition of the listening test always produced the same result.

This was no surprise to me, since I have been involved with engineers testing audio circuits by listening to them for 40 years (as well as measuring them, of course) and have learnt to be dispassionate, even to the point of rejecting circuits that theory and measurements said should sound better, because they didn’t. This was usually accompanied by a revision of our theoretical understanding and measurement techniques as we sought to align ourselves better with the real-world behaviour of the human auditory system.

Table 1.

<table>
<thead>
<tr>
<th>Frequency bands produced extra spectral lines</th>
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<tbody>
<tr>
<td>7.5kHz, 15kHz, 30kHz, 65kHz, 130kHz, 255kHz</td>
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Table 2.

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<tr>
<th>Power supply</th>
<th>Output rails</th>
<th>Noise (dB)</th>
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<tbody>
<tr>
<td>Linear</td>
<td>±18V</td>
<td>10.15</td>
</tr>
<tr>
<td></td>
<td>±13V</td>
<td>12.65</td>
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<tr>
<td></td>
<td>48V</td>
<td>14.75</td>
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Table 3.

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<th>Frequency pairings that produced extra spectral lines</th>
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<tr>
<td>7.5kHz, 15kHz, 30kHz, 65kHz, 130kHz, 255kHz</td>
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Mr Fourier’s famous transform allows us to evaluate a short segment of real time (the time domain) in terms of frequency response (frequency domain). The FFT of a single frequency injected into the input of an electronic circuit should only show a single spectral line at the input frequency. However, if the electronic circuit changes the input signal in any way, the FFT will consist of a series of narrow spectral lines that will be harmonics associated with the input frequency and the way in which the input frequency interacts with the electronic circuits. This means that an FFT can reproduce an alternative view of distortion to the input signal that is applied by the electronics of the device under test at its output port.

The FFT of a single frequency showed up the distortions in the electronics pretty well, but differences in the audio output with respect to each power supply unit proved to be a little too subtle to demonstrate a convincing result. If there were any intermodulation products between the PSU noise and the signal, they were well below the noise floor. Intermodulation product distortion is non-harmonic, and so much more audible than harmonic distortion, so it is possible that it could be below the noise floor but still audible. One way of detecting whether such IPs exist is to reduce the resolution bandwidth so the random (white) noise floor drops to a level that reveals the systematic noises like IPs and harmonics that were hiding in the grass. Unfortunately, such scans of any wide frequency range can take a very long time, and I was not able to carry out such tests in the time available. I needed to find an alternative method.

As it turns out, the Audio Precision AP2 is capable of analysing two frequencies at the same time. This is referred to as ‘2 tone testing’. The outputs of the two Audio Precision oscillators are combined using a passive mixing network and then applied to the input of the device under test. The output FFT will contain the two input frequencies plus any harmonics and intermodulation products generated by nonlinearities in the circuit under test.

Figure 3 shows the FFT response of the AP2 connected to itself. Only two frequencies are displayed. This means that the AP2 analyser circuitry does not add any distortion to the passively mixed oscillator outputs. But when the passively mixed oscillator outputs were connected via the mixing console, the output FFT was much more complex whenever power supply unit was connected. All pairs of frequencies selected produced additional spectral lines related to the two input frequencies. Lower frequency pairings (1k and 2k, 2k and 3k, ...7k and 8k) only produced extra spectral lines at harmonics of the input frequencies. But when one of the frequencies was above 8kHz, many pairings produced additional spectral lines when powered by the switch-mode unit. Of particular interest to me were some additional spikes between 8kHz and 13kHz — just the sort of frequency range that I would expect to affect the stereo image changes noted in the listening tests — plus the regular appearance of a spike at 9.8kHz.

Figure 4 shows the output FFTs from passively mixing 8kHz and 15kHz. FFT 4(a) is the result from the mixer powered by the linear unit, and FFT 4(b) when the switch-mode unit was providing power. The two traces are similar, except for the additional spike at 9.8kHz in the switch-mode trace. The harmonics of the input frequencies are clearly marked, but the 9.8kHz is not an intermodulation product of either 8kHz or 15kHz.

In the additional FFT Figures 5 and 6 (only switch-mode results shown), the 9.8kHz is very prominent again. This must mean that 9.8kHz is an intermodulation product between either f1 or f2 and the switcher noise. But if that is the case, why did it not show up in the single frequency tests?

To be absolutely sure of the results obtained so far, I need to repeat the single frequency FFT tests, making sure that the input level of the single tone test is the same as the overall input level of the two tone test. What I had overlooked was that with two tones at once, we get beating at the difference frequency, and that at the peaks (the maximum level of the combined signals) the level will be 6dB higher than in the single tone test. I was not able to check this before the mixer and switch-mode power unit used in the tests got returned to their owner. I should also have measured the noisy rail switcher’s main noise frequency to establish that it must be intermodulating with some of the test frequencies. To do a proper job, I need to express the additional noise spikes as intermodulation products — Fswitcher +/-nf1 or +/-nf2. I guess this means that there will need to be a Part 3.

REFERENCES

Essential reading for audio designers: