EQUIPMENT REVIEW

MUSIC LABS DP101 DIGITAL PROCESSOR

To the best of my knowledge, the Music Labs Digital Processor is the first digital-to-analogue converter to be designed and manufactured completely in Australia. This is bad news for any potential Australian competitors, because Music Lab's D/A converter is so good that everyone else will have to go back to the drawing board. It's also bad news for the imported competition, because the Music Labs DP101 is not only better than almost all of them, it is also considerably less expensive!

To say that the Music Labs DP101 Digital Processor is 'good', is an understatement. I just could not fault it. The chassis is impressively solid and the front panel is well laid out and of superior quality (5 mm solid aluminium plate). The circuit layout is beautiful (possibly the best double-sided, dual ground plane PCB I have seen), and all the components used in the processor are of the highest quality. The overall standard of workmanship equals or exceeds anything I have ever seen in any imported product—and that includes a host of world-renowned names.

The Equipment

As you can see from the photograph, there is not a lot to the front panel of the DP101. The left-most switch is for power, which comes into the unit via an IEC-standard 240 V Euroconnector. The switch adjacent selects the input to the converter—digital via optical fibre (a standard Toslink connector) or digital via co-axial cable (a gold-plated RCA connector).

The two switches at the far right are for switching the absolute phase of the output signal, and for muting it. The inclusion of phase switching is an excellent concept. Many CDs are incorrectly phased, yet few CD players (or digital-to-analogue converters for that matter) will allow you to correct it. It sometimes seems as thought there is a conspiracy amongst certain manufacturers to hide any potential faults in the CD system. After all, we can't have the general public thinking the system is any less than perfect, can we?

Above the switches is a row of coloured LEDs labelled, from left to right, Power, Co-Axial, Optical, 32 kHz, 44.1 kHz, 48 kHz, De-Emphasis, Phase and Mute. As you can see, the Music Labs Converter is not restricted to standard CD-style digital input signals. It can also handle digital satellite broadcasts, professional DAT, and DCC formats. It uses a Yamaha YM3623B chip to lock onto the incoming digital signal and strip off the desired digital data, which it re-clocks into an eight times oversampling digital filter.

This digital filter drives two of Burr-Brown's most recent devices, the 20-bit PCM63P-K converters, which use that company's latest co-linear process to perform the required conversion. One potential problem with this new BB converter is that it is so accurate that digital jitter must be less than 20 picoseconds if one is to avoid time domain errors. Music Labs solves this by using unique anti-jitter circuitry in combination with seven different power supplies (four of which are completely discrete), all driven by a custom-wound, Tortech toroidal transformer.

While we're talking about the circuitry, the digital and analogue sections are completely isolated by fast optical couplers. These couplers prevent digital noise from getting into the sensitive analogue circuitry.

Although switchable muting is provided, the unit also mutes automatically when there is no digital data present at the input. Both de-emphasis and phase inversion take place in the digital domain, thus neatly avoiding the possibility of errors creeping in at the analogue stage.

Why A Separate DAC?

Part of the reason for owning a stand-alone digital-to-analogue converter should already be obvious. Music Labs has gone to great lengths to avoid interaction between the digital and analogue sections of its converter. You can just imagine the problems presenting CD player manufacturers that have both the CD transport *and* the DAC in the one cabinet! The entire hi-fi world is united in

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agreeing that for superior electronic performance and superior sound quality, external digital-to-analogue converters are the only way to go.

But there is also a sound economic reason for owning an external digital-to-analogue converter. Most readers of Australian HI-FI probably have an 'old' CD player—and by 'old', I mean anything manufactured more than 12 months ago. Such players have inferior digital-to-analogue converters, which additionally suffer by being encased in the same box as the CD transport. But, although the internal DAC might be old, there's usually nothing wrong with the transport or the laser optics. (I say 'usually' because, to my knowledge, at least two multinationals have had problems with their bearing assemblies.)

By adding an external converter, a CD-player can be immediately upgraded. But there's more to it than that. If you have an external converter, you don't need to use the internal converter in your DAT or DCC player. Instead, you can use the superior conversion of the external unit—particularly if you are using a DAT or DCC portable.

Circuit Philosophy

Designer John De Sensi works with one guiding philosophy, that the less componentry there is in the way of a signal, the better that signal will sound. Accordingly, he told Australian HI-FI that the DP101 uses only the finest components he could find or, in the case of the transformer, have made, and has the fewest components in the signal path. One glaring example is the omission of post-filtering, which is actually recommended by Burr-Brown. According to De Sensi, he built two identical prototypes, one without post-filtering and the other with filtering. 'The DP101 without the filtering sounded better' he said.

'All the DACs I have seen use active filters, mostly with inferior-grade ICs' he said. 'And, even if a manufacturer were to passively filter to avoid using op-amps, you'd still need capacitors and chokes, which would result in clearly-audible frequency-domain errors.' Says De Sensi: 'Most manufacturers forget about the obvious—that anyone using a DAC will have to amplify the signal before it becomes useful, and almost all amplifiers manufactured are actually band-limited, which makes them quite effective filters in themselves. So why add additional filtering?'

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Serial Number: Not Stated

Warranty Period: Five Years

Price: $1,799.00

Distributor:

For further information about the Music Labs DP101, turn to page 82 and circle 0260 on our Reader Information Service Card.

Listening Sessions

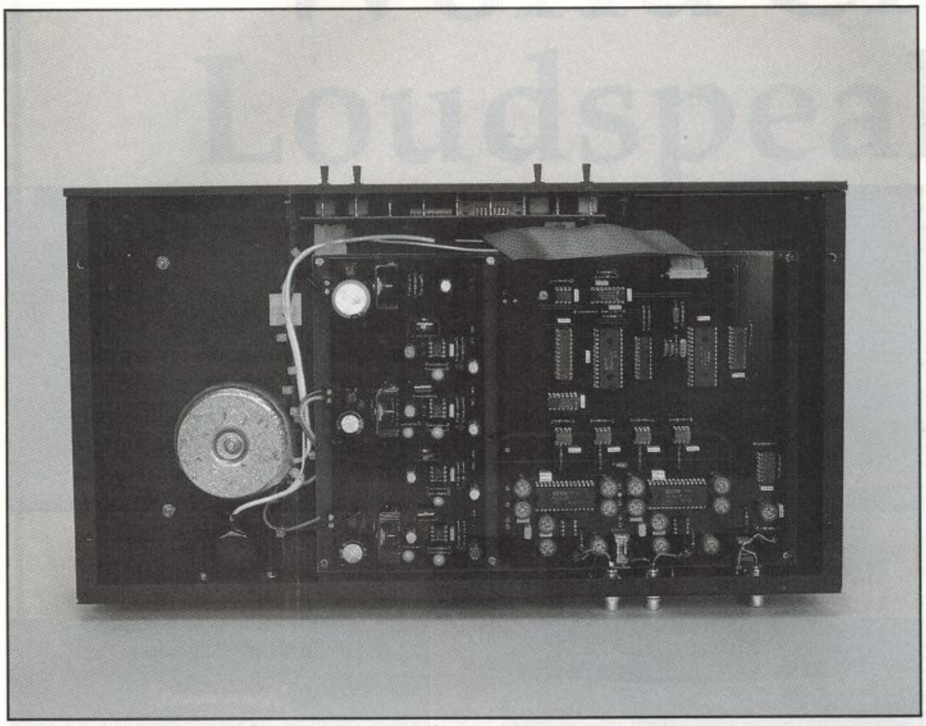
I must admit that I went into the listening sessions expecting to hear good sound—the evident thought that had gone into the circuit design and the execution of the unit itself was more than sufficient to inspire one's confidence. I also had the benefit of having the designer's own hand-made sample unit, no doubt previously tested and completely optimised. But even all this ground-work left me unprepared for just how good the Music Labs DP101 sounds.

Its sound was completely effortless, with an absolute lack of graininess, particularly at low recorded levels, yet the system did not strain when 0 dB transients came powering through, and retained exact tonal character. The fluidity of the sound was particularly noticeable, audible as a feeling of 'ease' when listening. Each note seemed to segue into the next, and each musical phrase slipped into the next with a total lack of jarring.

Whereas some converters can tend to sound rather two-dimensional, the Music Labs was decidedly three-dimensional, reproducing an almost-holographic sound stage that had equal amounts of height, depth and width. And, once the image was created, it became fixed in space, rock-solid, completely independent of frequency. The image was so precise that it was easy to detect which performers moved their instruments as they played—an effect most noticeable on trumpet and trombone, but clearly audible even with more overtone-free instruments, such as flute and piccolo.

Frequency balance was almost perfect. At times I imagined a slight roll-off in the bass, but at other times I couldn't fault it. The high frequency range is transparent, and achieves this without recourse to artificial roll-off or deliberate and quite artificial inter-channel phase shifting (a practise employed by at least one high-end manufacturer).

Although all the listening was done with 44.1 kHz signals, I listened briefly to signals clocked at 48 kHz and 32 kHz and could detect no anomalies. I therefore assumed that the benefits bestowed by the DP101 at 44.1 kHz would apply at these other sampling rates.



Conclusion

I was very pleased and encouraged to see that Australian designers and manufacturers can come up with the digital goods in world that's rapidly becoming exclusively digital. It is reassuring to find that we are capable of producing such high-quality components as the Music Labs DP101. However, I didn't let any of my patriotic (or parochial) thoughts interfere with my assessment of the product. After all, if you're spending $1,800 on a DAC, you'd expect it to be up to scratch. In my opinion, the DP101 is more than just 'up to scratch', it's a magnificent achievement. ■ *greg borrowman*

Readers interested in a technical appraisal of the performance of the Music Labs DP101 Processor should continue on and read the 'LABORATORY REPORT' published on the following pages. All readers should note that the results mentioned in the body copy, tabulated in the performance chart and displayed in the various graphs and photos should be construed as applying only to the specific sample tested.

Laboratory Report

We have published frequency response runs for both the left and right channels. There were small (vanishingly small) differences between them. For example, at 10 Hz, the left channel was down –0.57 dB whereas the right was down –0.64 dB. (The woodles on the trace are a function of the measurement technique.) Between 10 Hz and 20 kHz the right channel was flat within ±0.05 dB and the left was flat within ±0.03 dB. As you can see, the frequency rolled-off dramatically after this point, to be –22 dB down at 25 kHz and 31 dB down at 30 kHz. This, however, was not so much a function of the DP101, which does not have filtering, as of the test signal, which came pre-filtered off a test CD.

At the same time frequency response was being analysed, the Lindos computer set-up also analyses phase. In the case of the DP101 it measured it as being close to perfect—a mere 1° in error at 40 Hz and exact above that frequency. Also absolutely precise was linearity—the DP101 is the most linear device we have yet tested on the Lindos, maintaining perfect level accuracy all the way down to –90 dB.

The Music Labs processor didn't appear to add any more noise to the signal than was present in the original. Australian HI-FI Test Laboratories measured 'A'-weighted noise as better than –92.5 dB, but this is within the realms of measurement error for the test disc in question.

Channel separation was superb—the best we have ever measured for any CD player or DAC, irrespective of price. As you can see on the appropriate spectrogram, even with the lower (left channel) trace artificially elevated by 30 dB there is no sign of signal breakthrough. Separation was better than –130 dB.

Total harmonic distortion (bandlimited, of course) was measured at several frequencies. At 40 Hz it was 0.0373%, at 100 Hz, 0.0191%, at 1 kHz, 0.0168%, at 6.3 kHz, 0.0137% and at 10 kHz it was 0.02%. Very satisfactory indeed. You can see what this level of distortion looks like by inspecting the spectrograms showing the audio spectrum from 0 dB to 20 kHz with a 0 dB, 1 kHz sine wave and from 0 Hz to 50 kHz with the same input signal. Other than the sine test signal (the peak at the left of the graph), there is nothing visible above –90 dB.

Because of the lack of post-filtering, the same is not quite true when the frequency of the test signal is increased. At the maximum frequency permitted by the CD standard (20 kHz) you can see there are some spuriae generated around the fundamental as well as one up at approximately 39 kHz. These spuriae would be quite inaudible, firstly because few people can hear above 18 kHz and secondly because they would be totally masked by the 20 kHz tone. This same signal is shown in the time domain in the relevant oscillogram, and the stepped appearance of the waveform is quite obvious. These steps, in part, account for the spuriae

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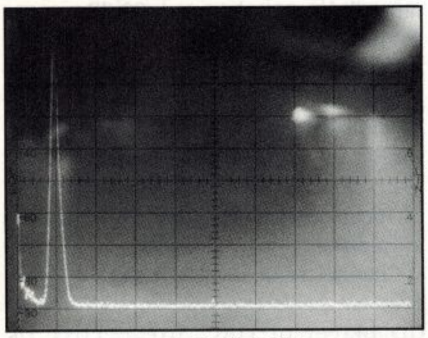
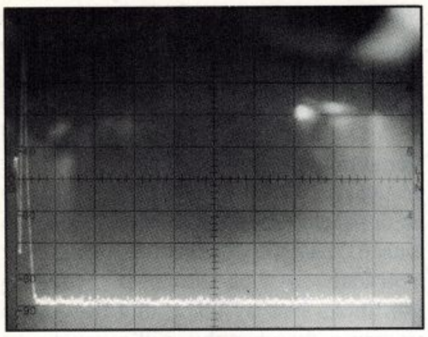
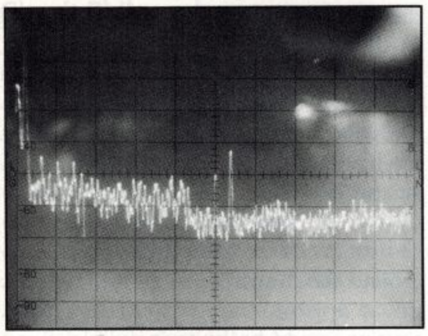


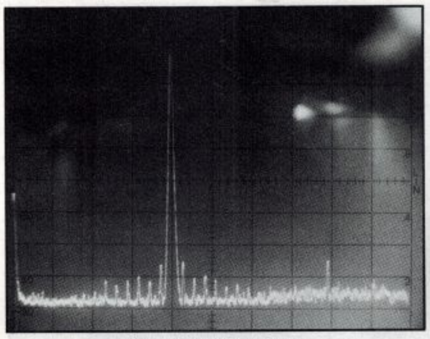
Distortion @ 1 kHz @ 0 dB
Horizontal 1 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



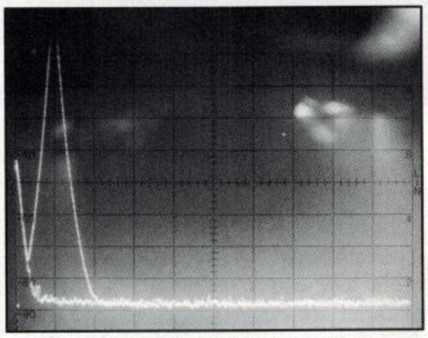
Distortion @ 1 kHz @ 0 dB
Horizontal 5 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



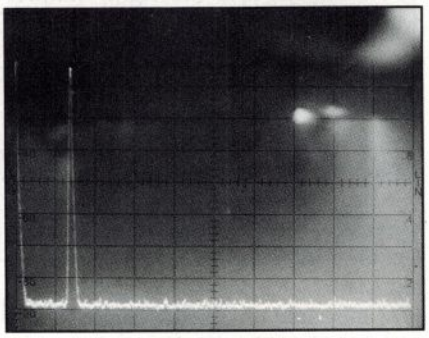
Distortion @ 1 kHz @ -60 dB
Horizontal 5 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



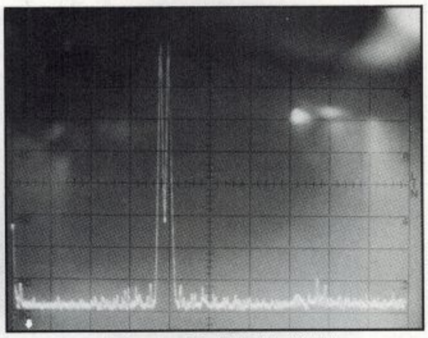
Distortion @ 20 kHz @ 0 dB
Horizontal 5 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



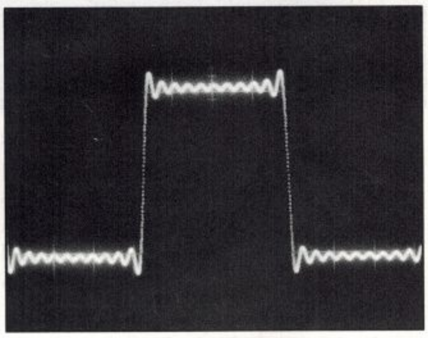
Channel Separation @ 1 kHz re 0 dB
Horizontal 1 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz
(Separation Trace Raised 30 dB)



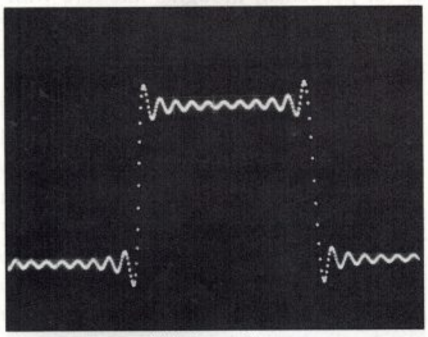
IMD Distortion @ 0 dB (60 Hz/7 kHz)
Horizontal 5 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



IMD Distortion @ 0 dB (19 kHz/20 kHz)
Horizontal 5 kHz/division; Vertical 10 dB/division,
Start Frequency 0 Hz



1 kHz Square Wave
Pierre Verany Test Disc
(See Copy)



1 kHz Square Wave
Philips Test Disc
(See Copy)

on the frequency analysis. The reason more isn't visible is simply that the frequency analysis stops at 50 kHz, so the 'distortion' generated by the steps is out of the Hewlett Packard analyser's passband (ie >50 kHz).

The DP101's lack of filtering was illuminating in that it reveals substantial differences between the various test CDs used by Australian HI-FI Test Laboratories. If you look at the oscillograms, for example, you will see we have shown four supposedly-identical 1 kHz square waves. As you can see, it appears that what is recorded on test discs are not necessarily 'true' square waves at all. Philips appears to have rounded-off the data it stores on its test

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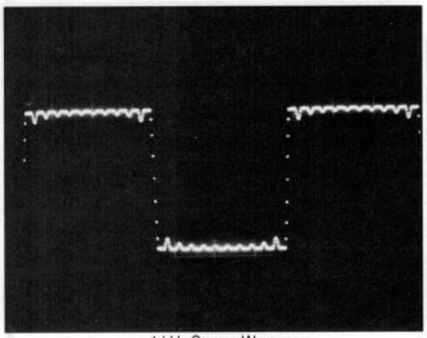
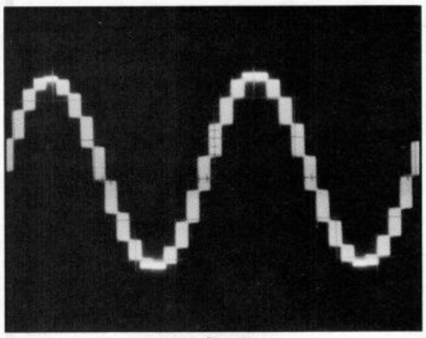


1 kHz Square Wave
CBS Test Disc
(See Copy)



20 kHz Sine Wave
(Showing Quantisation Steps
due to lack of filtering)

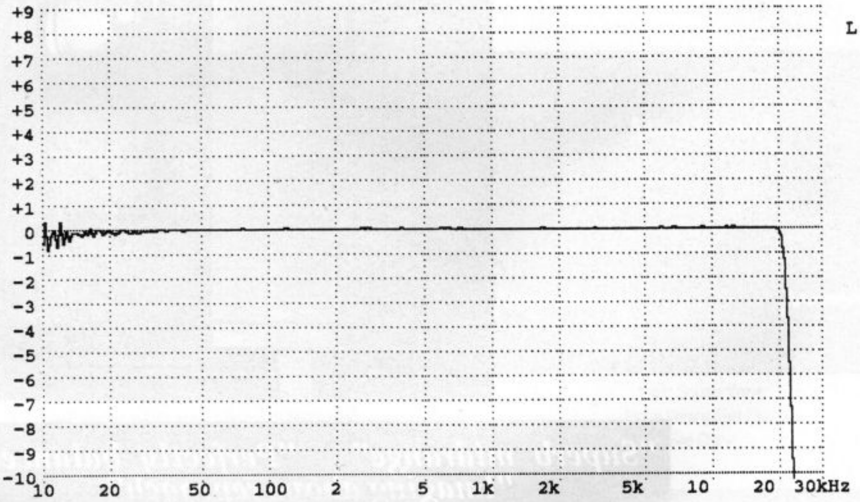
disc, while CBS appears to have used Sony's new oversampling procedure for recording, which apart from resulting in the peculiar waveform shown, also inverts the signal. The Pierre Verany version shows the most 'recorded samples' and is thus, technically, 'the best'. Ultimately, the shape of the final waveform entering your speakers will depend on the post-filtering applied by your amplifier. British amplifiers, for example, which often roll-off before 30 kHz will provide greater filtering than Japanese and American amps, which usually don't start rolling off much before 50 kHz. Ultra-wide-band amplifiers (those whose responses are measured in hundreds of kHz) would, of course add no filtering to the signal.

Low-level conversion accuracy is shown on the wideband –60 dB spectrogram. As you can see, the noise floor is at or better than –100 dB below 20 kHz then takes a sharp drop above this frequency, to be at or better than –120 dB. The two small 'peaks' visible on the spectrograms appear to be artefacts of the new Burr-Brown process, and are located at approximately 25 kHz and 27 kHz.

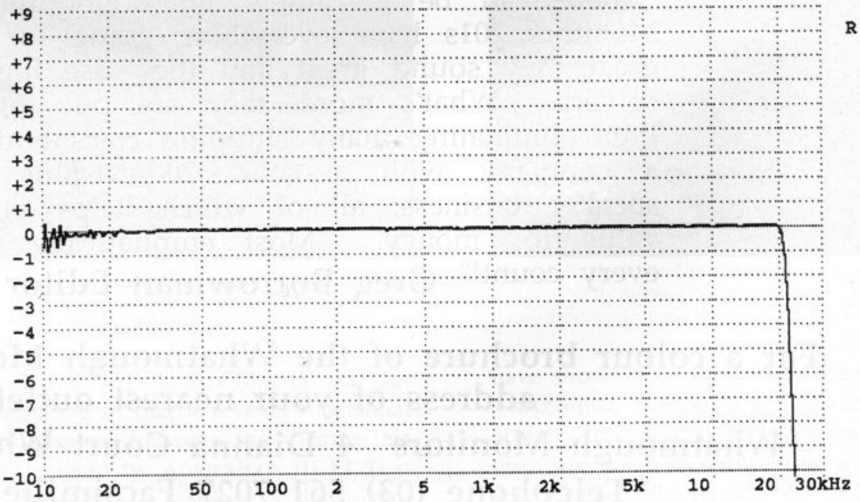
Intermodulation distortion testing revealed no anomalies. In fact the twin-tone 19/20 kHz test signal appeared to *reduce* the spuriae which appear when only a single high-frequency tone is applied. This bodes well for music signals, which are never simply sine in form, since it indicates that more-complex waveforms actually cause the DP101 to behave better.

Output voltage was measured at 1.81 mV so the DP101 will interface easily with any pre or power amplifier. ■

greg borrowman



Music Labs DP101. Frequency response at 0 dB. (Left channel).



Music Labs DP101. Frequency response at 0 dB. (Right channel).