Studio Sound, December 1991

A FUTURE OF MIXING?

Concluding our retrospective look at the Yamaha DMP series, Simon Sanders covers the digital-only models.

Last month we looked at the *DMP7*, an 8-channel line mixer, offering fully automated mixing and processing, with in-built parametric equalisation and multi-effects processors. At the input stage, the audio signal is converted from analogue to digital and all internal processes are carried out in the digital domain. The digital signal is then converted back to analogue at the stereo output, unless you are using the FMC1 to convert the proprietary Yamaha digital signal to the Sony or AES/EBU formats, in which case D/A conversion is only necessary for monitoring purposes.

Offering identical processing facilities, automation capabilities and even outward appearance, the *DMP7D* could very easily be mistaken for its sister machine. But, if you were to attempt to plug an audio lead into the *DMP7D*, you would be hard pushed to find a channel input socket on the rear panel, for the simple reason that there aren't any. This is because the *DMP7D* dispenses with analogue inputs altogether. Instead, the unit will accept input from almost any digital source - be it multi-track, stereo or even multiple sources.

Digital input

On the rear panel of the machine there is a 25-pin DSUB connector for input of up to eight channels of digital audio. Input on the DSUB connector is switchable between Yamaha, Mitsubishi and Sony interconnect formats. The DSUB connector cannot be used for direct input, but requires the use of an external interface unit.

There are five interface units available at the moment, (*IFU 1, 2 3* & *4* and the *IFUD2*), which are supplied as lU rack units, except for the *IFU1* which is 2U high. The *IFU1* is designed to take the two 50-pin outputs from a Mitsubishi *X850* 32-track PD-DASH machine and distribute it to the 25 -pin inputs of four *DMP7Ds*, (which are then cascaded using the digital cascade format). When using the *IFU1*, the input format selector should be moved to the 'M' position, (Mitsubishi format).

The *IFU2* is designed to take the 50-pin output from a Sony *PCM-3324* and distribute it to three *DMP7Ds*, with the input format selector switch in the 'S' position, (Sony format). The *IFU3* is designed to route the left and right outputs of four Sony *PCM-1610/1630s* to a single *DMP7D* and the *IFUD2* routes four AES/EBU format signals to a single *DMP7D*.

The *IFU4* serves a slightly different purpose, in that it converts input signals from TTL level to RS422 level and can be used to distribute up to four clock signals for synchronising the data streams of several sources. When using the *IFU1*, *2* and *3*, one or more *IFU4(s)* may also be required.

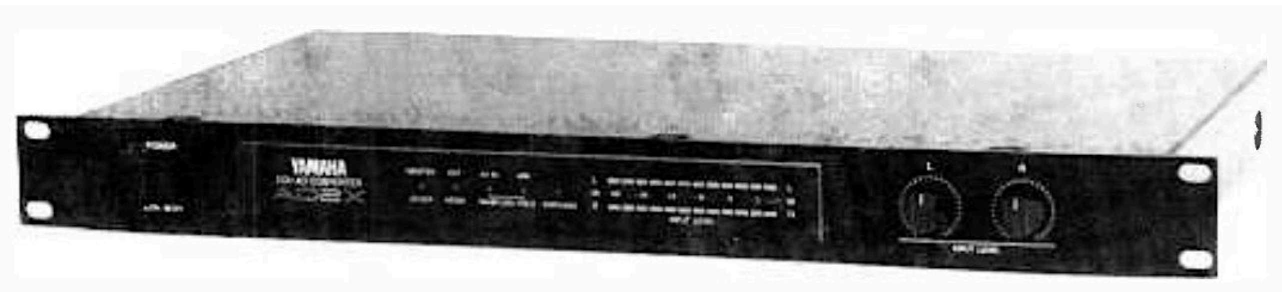
The rear panel of the *DMP7D* also allows for direct input of one AES/EBU format or one CD/DAT format signal. The AES/EBU or CD/DAT signal is automatically routed through channels 7 and 8 of the mixer, overriding channels 7 and 8 of the 25-pin input, and automatically switching the DMP7D to external word clock, (using the self-clocking AES/EBU signal for synchronisation).

Analogue inputs

Analogue input to the *DMP7D* is possible through the use of the *AD808*, an eight channel analogue to digital converter. A/D conversion is achieved with the same successive approximation converters as used in the *DMP7*, giving a 16-bit linear quantised signal.

The unit has eight analogue inputs, using balances XLRs at +4dBu for 600 ohm lines. The *AD808* is connected to the *DMP7D* via the 25-pin input and the input format selector must be moved to the 'Y' position (Yamaha format). The front panel has eight peak hold LED meters and eight rotary input level controls. The rear panel has switches for emphasis on/off, sampling frequency, (44.1/48 kHz) and word clock source selector. Signal synchronisation can be achieved either internally, via the 25-pin connector, or via an external clock source (in the Sony word clock format at TTL level). The status of each of the above functions is indicated on the front panel by a series of LEDs. The AD808 is again rack mountable, occupying 2U of rack space.

If you only wish to input two analogue channels, it is possible to use the *AD2X*. The *AD2X* is a 19-bit analogue to digital converter, using 1-bit delta-sigma converters with 64 times oversampling. As with the AD808, analogue input is via balanced XLR connectors at +4 dBu. Digital output is available in three formats - Yamaha MEL2, AES/EBU and CD/DAT. There is a word clock input/output, enabling the unit to act as master clock source or as a slave. Emphasis can be switched in or out and the sample rate is selectable between 44.1 and 48 kHz. The front panel has two peak level LED indicators and input level controls.



The output of the *AD2X* is of high quality. The use of 19-bit conversion gives potential dynamic range of over 110 dB. The *AD2X* can be used as a stand-alone converter for use with DAT recorders or other digital recording systems.

The *AD2X* can be connected to the AES/EBU or CD/DAT inputs on the *DMP7D*, or alternatively, to the Yamaha cascade input, effectively acting as an extra pair of input channels, (however, using the cascade input will not allow you access to the internal effects and EQ). This particular configuration could be used to route the output of a conventional console directly to the stereo bus of the *DMP7D*, allowing you to mix analogue sources alongside digital sources.

If you are using analogue sources on a regular basis, it may be a better option to use a *DMP7* connected to the cascade input of the *DMP7D*, thus giving the opportunity to mix both analogue and digital sources within the same mixing system.

Digital outputs

Output from the *DMP7D* is available in several formats: AES/EBU format via a single XLR connector; CD/DAT format via a single RCA jack; SDIF via either a pair of BNC connectors (TTL level), or a pair of XLR connectors (RS422 level). There is also a Yamaha MEL2 output (via an 8-pin DIN connector), for cascading to another *DMP* unit, (*DMP7*, *DMP11* or *DMP7D*).

The *DMP7D* does have a pair of analogue outputs, which are intended for use as a monitor output only, as the quality of the D/A conversion is not particularly good when compared with some of the other converters available. There is also a stereo headphone socket provided. Monitor level can be adjusted with a small rotary pot located on the rear panel - the master stereo output cannot be used to alter monitor level without affecting the digital output level.

For high quality digital to analogue conversion the AES/EBU or CD/DAT output of the mixer can be connected to the *DA202* which offers 18-bit D/A conversion with an 8 times oversampling filter. Analogue output is via a pair of balanced XLR connectors at +4 dBu. The unit also provides a digital thru output, so that a DAT machine, for example, can be used to record the digital output, whilst analogue output is recorded on an analogue recorder. Like the *AD2X*, the *DA202* can be used as a stand-alone unit.

Word clock

On the rear panel of the *DMP7D* is a three-way selector for word clock source. In position 'A', signal synchronisation is taken from the cascade input, in position 'B', from the word clock input, (in the Sony word clock format at TTL level) and in position 'C', from the DSUB connector.

For signal synchronisation of external devices, there is a word clock output, again in the Sony format. The cascade output also carried a word clock signal to synchronise with other Yamaha devices to which it is connected.

When using the interface units, signal synchronisation can be quite a problem, and this is where the *IFU4* becomes necessary. If using the *IFUI*, the internal clock output of the Mitsubishi will need to be connected to an *IFU4*, with the four RS422 outputs connected to the four word clock inputs of the *IFUI*, (which then routes the clock signals to the DSUB connectors of the four *DMP7Ds*). In addition, the *IFUI* has a bit clock output which will need to be connected to another *IFU4*, with the four RS422 outputs connected to the four bit clock inputs of the *IFUI*, (again for routing to the four *DMP7Ds*). *Confused*?

I feel that Yamaha could have simplified matters substantially by incorporating the clock distribution into the casing of the *IFUI* and just having a single word clock connection from the Mitsubishi to the interface unit - it may well have cost more, and the unit would have occupied more space, but surely no more so than the one *IFUI* and two *IFU4s*. It would also have reduced the apparent complexity of the system and cut down on the external wiring requirements. (The same points apply to the *IFU2*, which requires one *IFU4*, and the *IFU3*, which requires three *IFU4s*).

Yamaha argue that the cost of incorporating clock distribution into the interface units would have made them too expensive and that they would still have had to produce the *IFU4* anyway. Also, producing the interface units as separate modules makes the system, as a whole, a lot more flexible.

Even with word clock being used to synchronise the data streams of the various units in a digital system, it is still possible for the data to be misread due to delays introduced by cable runs, conversion processes etc. This is not usually too much of a problem as we are normally only dealing with a stereo signal, (which in the case of the AES/EBU format, is sent as a single self-clocking data stream). But, if you are using a digital multi-track, data delays can be a major headache when it comes to laying material in sync with that already on the tape.

To account for these delays, the *DMP7D* has a 'bit-shift' facility, in the form of two rotary controls on the rear panel, (fine and coarse adjustments). These can be used to synchronise signals being sent to the recorder with those being monitored off tape. Setting the bit-shift can be done by ear, but is most effectively done with a scope - something best left to a technician!

Data synchronisation is an area where there is a lot of confusion, especially in systems which are using many different digital recording and processing devices. The BBC Radiophonic Workshop have had many problems with data synchronisation in their digital audio system, but are often unable to isolate the problem to a single unit - it tends to be a 'system' problem.

External effects

Like the *DMP7*, the *DMP7D* has three internal effects processors, and again, effect send three can be used as an external effects send. The effects send itself is located on the rear panel, and consists of a digital output in the Yamaha MEL2 format. The effects return is also in this format.

The external send can be used with the Yamaha *SPX1000* or *DEQ7*, by simply plugging in two 8-pin DIN leads, giving access to very high quality multi-FX or digital equalisation with minimum fuss.

However, using an alternative external effects processor introduces problems, as the Yamaha format digital signal must be converted either to another digital interconnect format, or to analogue. It is possible to use the *FMC1* to convert the Yamaha MEL2 to AES/EBU or SDIF for connection to, for example, one of the Roland digital effects processors, although there is no box, as yet, to convert AES/EBU to Yamaha MEL2 for returning the signal on the digital effects return - the AES/EBU return must be input using the AES/EBU interface for channels 7 & 8, or by using the *IFUD2*.

If you wish to use an analogue signal processor, then you must use the *FMC1* to convert the Yamaha signal to AES/EBU, and then use the *DÁ202* to convert the AES/EBU signal to analogue. The return signal must then be fed into the *AD2X* which is, in turn, connected to the effects return input.

If you are going to be using an *SPX1000* or *DEQ7*, then the external effects send is an excellent facility, as you have access to high quality digital processing without converting your signal to analogue and back again unnecessarily. Using other digital processing devices, you may have to start using format converters to keep your signals in the digital domain, but if you consider the sonic advantages of this, it may be worth your while to invest in these format converters. If you want to use an analogue signal processor, then outboard D/A and A/D converters are needed and may be more bother than they are worth, (although if you are not using the effects send for an external processor, it can always be used as a monitor mix send (mono), whilst the return can always be used as an extra stereo input channel).

Conclusion

There are many engineers in the industry who have not yet begun to consider the possibilities that are opening up in the digital arena. With tapeless recording /editing systems and DAT becoming commonplace, and with timecode compatibility being incorporated into these systems it will not be very long before the audio industry accepts digital recording techniques as standard practice. When this happens, people will start to realise that it is counter-productive to convert the high quality digital audio signals to analogue, and then mixing them through an analogue console before converting them back to digital for mastering.

In audio for visual applications, the use of digital recording and editing systems offers a wealth of creative possibilities, especially with the introduction of NICAM stereo in European broadcasting. The use of DAT for location sound, CD libraries for sound effects, direct digital mastering of dialogue and the increased use of CD or DAT for the music, means that the A/V engineer is going to have to seriously reconsider the mixing aspect of the post-production process.

The *DMP7D* is, at the moment, the only 'affordable' digital mixing system which can be used to mix digital signals without leaving the digital domain. It offers high quality equalisation and effects, as well as full automation. It is extremely flexible regarding the interface formats of input signals, (as long as you are prepared to put up with a pile of 'black boxes' between the various units). The *DMP7D* can also be fully integrated with the *DMP7* and 11 using the cascade I/Os, and is totally compatible with the *RTC1* for remote operation.

One of the few problems with the unit is that the input configurations are fixed. For instance, there is no way of inputing four channels of analogue using the *AD808*, whilst inputing four channels of AES/EBU format digital signal, (although there are ways to get around this, as always in the wonderful world of audio!).

Regarding its use with digital multi-track machines, I reckon that any studio which is going to invest the huge sums of money involved in purchasing the multi-track machines, will not feel happy about using the *DMP7D* if only because of the appearance, (their clients will expect to see a more conventional console). This is not to say that the *DMP7D* is not suitable for multi-track applications.

As to the future, we are waiting for the manufacturers of synthesisers and samplers to start including a digital output on their machines. When this finally happens the *DMP7D* will come into its own and will become a necessity for any serious MIDI musician. All in all, the *DMP* series offers the audio world a combination of facilities that are difficult to beat in terms of affordability and compactness. They represent an extremely flexible and integrated approach to the mixing of material from both analogue and digital sources. Used with the *RTC1*, the *DMPs* can be used in a 'desktop' situation, obviating the need for large premises to house enormous consoles. Some day all mixers will be made this way!

Author's note: *I would like to thank Yamaha-Kemble (UK)*