Neumann Solution-D

The digital microphone. From its inception the concept has always seemed to be a strange contradiction in terms, and an attempt for the digital business to infiltrate one of the last analogue bastions in the audio chain. Yet you can see the merits of the idea, says JON THORNTON.

GIVEN THAT AN INCREASING number of microphone signals these days find their way straight out of a preamplifier and into an A-D convertor, there is a logic in trying to move this conversion stage as close to the analogue transducer as possible. And in this case, as close as possible means immediately after the capsule of a capacitor microphone, dispensing with the usual analogue amplification stage in the microphone body.

At this stage a fundamental problem occurs. The dynamic range from a good capacitor diaphragm is typically of the order of 130dB. The effective dynamic range of commercially available 24-bit A-D convertors is roughly 115-120dB, and therefore still presents something of a bottleneck in the pursuit of the (as near as possible) digital microphone.

This was the challenge presented to the design team at Neumann when creating the Solution-D, which is in truth more of a digital microphone system as opposed to a standalone microphone. Although the Solution-D isn't exactly a new product, it was first announced in 2001, it is only since last September that production models started to ship in any quantity. And given that it is still a relatively uncommon beast on these shores, the chance to review not just one microphone but a stereo pair was too good to pass up.

Solution-D consists of one or more Neumann D-01 mics, together with a DMI-2 interface unit for each pair of mics, and some very nifty remote control software, of which more later.

Returning briefly to the A-D problem, Neumann has come up with a rather elegant solution that involves mapping the signal from the capsule to two separate convertors. One of these handles high level signals and the other lower level signals after they have been passed through an amplification stage that drives low levels into the convertor across its whole range. While there is nothing new in this 'gain-staging' approach, in the past implementing it has been fraught with difficulty. For example, because the convertor handling the amplified low level signals is mapped in such a way that it will inevitably be driven into clipping as the signal level increases, the digital output must be able to detect this point and switch seamlessly to and from the high level convertor without audible artefacts. This is harder to achieve than it sounds, as exceptionally small timing errors can have significant audible effects.

The Neumann approach tackles this from the other end of the chain, and somewhat perversely employs analogue signal processing to modify the signal levels presented to the two convertors, so that neither is ever over-modulated. In essence this is achieved by using a non-linear network that progressively discriminates against low level signals, the output of which is used to feed the high level convertor and to attenuate the level of the low level signal as it is amplified. The outputs of both convertors are then simply added together by DSP (after having first digitally attenuated one of them by the same factor as the amplifier) to produce the final output. Clearly, this is merely a thumbnail sketch of the process and interested readers can follow this up in Stephan Peus and Otmar Kern's paper to the 110th AES Convention. The really clever part is that the circuit topology means that any artefacts caused by the analogue network are effectively cancelled out at the digital addition stage.

Neumann claims an effective dynamic range for this convertor of 140dB, equivalent to an internal resolution of 28 bits, and a dynamic range for the convertor and capsule of at least 130dB (A-weighted).

The D-01 mic itself is a substantial looking but reasonably compact package, which doesn't really give much away. Housing a newly developed double 30mm diaphragm capsule, there are no external controls. The front of the mic sports two rectangular LEDs, and that's it. The only thing that hints that this is anything other than a run of the mill fixed-pattern condenser, is that the conventional XLR connector is labelled AES42-2001.

This marks another technical milestone, which is Neumann's adoption of, and input to the creation of, this standard. AES42 differs from the conventional AES-EBU digital interface standard in a number of ways. First, it allows for transmission of status data as well as audio data from the mic. Second, it includes the capability to include common mode signals to be delivered upstream for carrying power and control data. Finally, it includes a method for overcoming another inherent problem in developing digital microphones – digital synchronisation.

This part of the AES42 standard was developed by Neumann, and allows accurate digital synchronisation to be achieved regardless of sample rate and cable length. Referred to as Mode 2 synchronisation in the standard, it is in principle a phase locked loop. The microphone itself contains its own voltage controlled crystal oscillator, which clocks the A-D convertor. Audio data from the mic is compared with a master clock at the receiver side of the connection, and a phase comparison is performed. This process is used to determine a control signal, which is then sent back up the AES42 link to modify the mic's internal clock rate, which Neumann claims leads to negligible jitter amplitudes.

In theory, the D-01 could be plugged into any digital console or other device that supports AES42 – in practice such devices are rather thin on the ground at the moment, hence the provision of the DMI-2 interface unit. This unit will accept AES42 inputs from up to two microphones, and outputs standard AES-EBU digital pairs. In addition, the DMI-2 can function as a master clock generator at sample frequencies up to 192kHz (although the microphone currently only supports sample rates up to 96kHz), and will provide this clock via the Mode 2 synchronisation method detailed above to the microphones, as well as making it available as a word clock source to other devices. Alternatively, it will accept incoming word clock from an external source to use for this purpose. An RJ45 connector from the DMI-2 can be plugged, via an in-line convertor, to the USB port of a PC. This allows the use of the third part of the system, the RCS remote control software.

The software – currently only available for PC (a Mac release is imminent) – is never going to win any awards for look and feel, but is certainly stable and functional. A virtual control strip for each microphone allows the selection of parameters such as polar pattern, and pre-A-D attenuation in 6dB steps down to – 18dB. This is achieved in the microphone by the use of a field programmable gate array, controlled via AES42.

Further down the channel strip comes a digital gain setting, allowing 63dB of gain in 1dB steps. Metering is provided at the bottom of the strip, together with buttons for phase reverse and mute. These are now performed at the microphone rather than at the console. The mystery of the two LEDs on the front of the mic is also revealed, as there are buttons to toggle a blue and a red LED on each microphone on and off – even parameters for altering their intensity. Potential uses for cue lights and mic active lights spring to mind. The intention is to enable external control inputs via a port on the DMI-2 to control these and other parameters on future releases.

Perhaps the most intriguing part of the control software is the inclusion of peak limiting and de-essing parameters built in. These functions are achieved within the DSP on the microphone, and enable severe transients to be dealt with as close to source as possible. Attack and release parameters are available for both, together with auto release time options.

Additional buttons appear in the middle section of each strip – some clearly labelled as inactive, but others labelled with noise-gate and EQ functions. While these are not implemented in the current software, it clearly indicates Neumann's future intentions. Delay compensation in individual samples is also selectable for each mic. Finally, individual mics can be named on the channel strip, and the mic type and serial number is automatically transmitted and displayed.

All of which is a lot of background before we finally get to the heart of the matter – how does it actually sound?

For a large diaphragm microphone, the D-01 sounds surprisingly neutral and flat – it's not without character. I couldn't find any published response curves, but there seems to be a very gently elevated high frequency response, without any obvious presence bumps, and a similarly well behaved but incredibly full low frequency response. What hits you immediately is the transient response of the microphone, which adds a feeling of openness and transparency that overshadows anything in the frequency response. Given that you know exactly what is going on inside the microphone, you find yourself listening very hard for any obvious processing or conversion artefacts – if they are there then I can't hear them.

In truth, you find that you don't actually have to use the pre-attenuation function much – which proves the point about the conversion process. The built-in limiting and de-essing function, if driven hard produces some unpleasant results, but used in moderation almost gives you the feeling that there are no transients that are likely to occur in a recording studio that the microphone can't happily deal with. It's a dangerous notion and probably one that in the wrong hands could lead to an over reliance on DSP to solve problems. In spite of all this though, while you don't walk away from the microphone dissatisfied, there are times when you wonder whether the quest for the digital microphone has taken some of the character out of a Neumann's sound.

And this, I think, is the point. Whether Neumann sells many of these mics is to some extent immaterial – although they deserve to. What is important is that Neumann has got an incredible 'proof of concept' here, and at the same time may have started to redefine the role of the microphone in a studio.

I don't think it will be too long before we start to see this technology trickling down to other levels of product – maybe not with all the bells and whistles that are possible, but certainly the core concepts. And if manufacturers of mixing consoles and DAWs start adopting the AES42 standard, much of the control of this and other mics could be implemented rather more seamlessly within these environments. Couple that with the DSP capabilities of the microphone, and the potential for onboard 'mic-modelling' to make such a clean source sound like whatever you want it to, and you start to see the potential.

And that it is Neumann, a company with such a heritage in the analogue domain, that has made the first big step into the digital world is telling in itself. Hat's off to them. ■

PROS Dispenses with the need for analogue mic preamps, external A-D convertors, etc; fantastic transient response; open, detailed sound; remote control of parameters; onboard limiting.

CONS Lack of direct interfacing possibilities with other devices; software looks slightly 'clunky'; sound may be a little too neutral for some die-hards.

Contact

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