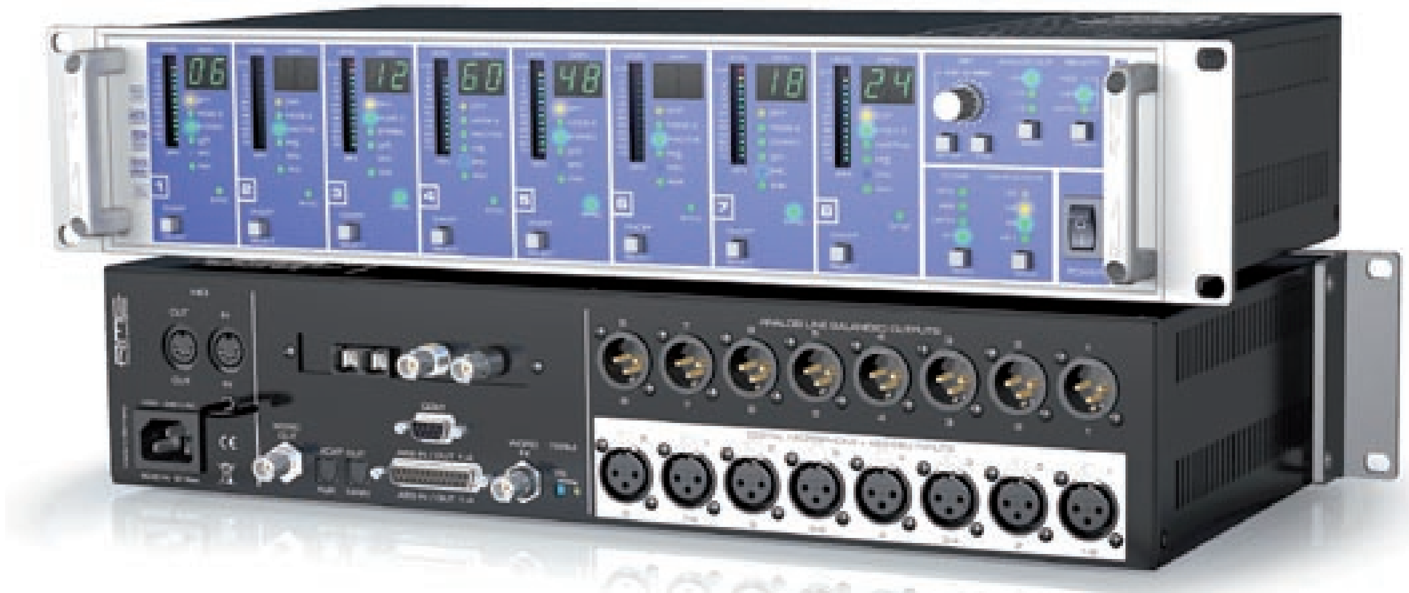
RME DMC-842 & Schoeps CMD 2U

After something of a shaky start, digital mics are with us and most importantly there's now a standalone preamp that understands and talks to them all. JON THORNTON does some of his best work in explaining the concepts and the operational principles.



It's taken a good while, but the adoption of AES42 as an interconnect standard for digital microphones seems to be gathering momentum. While Neumann has developed its original Solution-D concept and packaged it in the smaller form factor of the KM-D range, Schoeps has entered the fray with its CMD 2U digital mic amplifier. As this is effectively just another mic body option in the already extensive Colette modular microphone range, it means that it can take advantage of all the existing capsules and accessories.

Unlike Neumann's approach, which involves putting an A-D stage immediately after the capsule, the CMD 2U features a standard preamplifier (sonically the same as the CMC series bodies), which then feeds an onboard A-D convertor. The unit supplied for review was the 'xt' variant, which features an extended frequency response (>40kHz), and can use any capsules from the Colette range that are axially, rather than side-, addressed.

Until now, microphone manufacturers that have experimented with digital microphones using the AES42 standard have supplied their own interface boxes, with varying degrees of software/hardware control and scalability. The RME DMC-842 is a first example of a third-party manufacturer designing an interface that supports the standard.

In essence, the unit functions as an eight-channel mic preamplifier with analogue and digital outputs. The difference, of course, is that the microphone inputs are all AES42 digital source inputs. So, a better description is a multichannel digital mic controller and D-A convertor. A quick glance at the rear panel reveals eight digital microphone inputs on XLR. These can also accept AES3 signal pairs if required, making it possible to use the unit as a straightforward D-A convertor. Analogue outputs are on individual XLRs, and digital output is available as standard as four pairs of AES3 on a 25-pin D-Sub connector, or as ADAT format on lightpipe. A secondary lightpipe output allows sample rates higher than 48kHz to be supported using S-Mux. An external Word clock input, with switchable termination, is provided on a BNC connector as is a separate Word clock output. The AES3 connector also looks at incoming AES3 inputs for external sync purposes only.

A MADI card is also available as an option for the unit, supporting either 56 or 64 channel I-O. This might seem like overkill for what is an 8-channel device, but multiple units can be cascaded together going from the MADI Output (coax or optical) of one unit into the MADI input of the next. Each unit adds eight channels of audio to the MADI stream, and the unused blocks of eight channels are passed through to the next unit. The MADI output of the final unit — up to a maximum of eight — will then contain all of the audio channels. A neat if rather expensive solution.

The front panel is dominated by eight identical strips, each featuring a LED level meter, a number of status LEDs, a single pushbutton and a two segment numeric display. In most cases, this display indicates the amount of (digital) gain applied to each of the inputs, ranging from 0 to 63dB. Most of the operations and parameter adjustments are made using a single rotary encoder. Repeatedly pushing this steps all eight channels through different parameters, indicated by the status LEDs on the channels. The relevant parameter can then be switched On or Off on a per channel basis (for example, digital phantom power) using the pushbutton on that channel. Or, in the case of gain adjustment, the switch selects the channel to be adjusted, and the rotary encoder sets the level. A nice touch with gain level adjustment is that if this function is selected, but no individual channel is selected, the encoder will ramp the gain level up or down across all channels taking into account any individual offsets already in place. It sounds a little fiddly, and it does take some getting used to, but after a while it's quite a straightforward and quick user interface.

As well as toggling Digital Phantom Power (DPP) on or off for each channel, other basic parameters include switching odd and even pairs to function in stereo mode, with the option of MS decoding. In this case, the even numbered display strip becomes inactive (shown by a LED), and gain adjustments are applied equally to both channels. Sample rate conversion of the incoming signal to the unit's current sample rate can also be toggled on or off.

This is an important function, as a microphone like the Schoeps CMD 2U is what you might term a 'no frills' implementation of the AES42 standard. Clock source for the A-D is always internal to the microphone (known as Mode 1), and the sampling frequency preset at the factory with the rate engraved on the mic body (192kHz in this case). In fact, other than the requirement to use DPP over AES42, it could simply be described as a microphone with an AES3 digital output. Clearly, mixing and matching different microphones with different clocks and potentially different sample rates requires a sophisticated and high quality approach to synchronisation and SRC.

The clock source for the DMC-842 can be sourced internally, from an external TTL Word clock source, or from embedded clock from an AES input or MADI input. RME makes a big deal of the clock circuitry employed by the unit, called SteadyClock. Essentially this makes use of high-speed digital synthesis, a digital PLL and ultra-high (100MHz) sample rates to provide highly accurate clock generation or refreshing. Initially designed to clean up and reduce the high jitter values found in embedded MADI clock, the system enables any external clock source to be designated as a master clock, refreshed and jitter reduced, and then passed on to other devices if necessary.

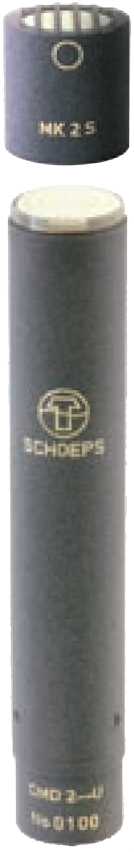
Sample rates of 44.1, 48, 88.2, 96, 176.4 and 192kHz are supported regardless of clock source, selectable from the front panel. Regardless of which clock source is chosen, a clever use of status LEDs for all available digital inputs also ensures that any potential problems with asynchronous clocks are also quickly indicated (if anything other than the current clock source is flashing, it's asynchronous).

In conjunction with a digital microphone, such as the Schoeps, operating in Mode 1 for sync, what we've described so far is nothing more than a very good sample rate convertor and D-A convertor, with the capability to alter gain digitally. To really get to grips with the capabilities of the DMC-842, you need to plug in a digital microphone that makes greater use of the capabilities of AES42.

Remembering that the standard allows bi-directional communication, this means that more is possible in terms of microphone control, and indeed allows a certain amount of DSP to be undertaken by the microphone itself. Perhaps most crucially, it allows some designs to make use of Mode 2 synchronisation if they support it. This means that, while the microphone's on-board clock still runs the A-D convertor, this clock can be controlled and locked to the DMC-842's clock source by sending control pulses up the AES42 link, alleviating the need for sample rate conversion.

The DMC-842 automatically detects a microphone that can run in Mode 2, and signals this with a status LED on the channel strip. The microphone (for test purposes a Neumann KM-D with a 183 omni capsule) will then synchronise its internal clock with whatever clock source and sample rate the controller is running at. In addition to synchronisation, this control data can also be used to remotely command a variety of functions and processes onboard the microphone. These include pattern selection, a low-cut filter, peak limiting, muting, and pre-attenuation.

Not all microphones support all of these functions, but the DMC-842 can control them if they are supported — again on a per channel basis. Pressing a 'Parameter' button under the rotary encoder enables it to scroll through these functions, which are somewhat cryptically displayed across the unit using the numeric displays. Holding down the select button on an individual channel then lets the rotary encoder select a numerical value for that particular parameter. It works, even if it is a bit clunky, largely because the values are simply numerical. You have to know, for example, that a value of '2' for pre-attenuation corresponds to -12dB. Similarly, there's no way of knowing from the front panel whether a particular function is supported by a specific microphone, so you can merrily alter the parameters with no effect. These issues are largely solved by the remote control software provided, which requires a PC running Microsoft's .NET framework version 2.0 or higher with a MIDI interface to communicate with the unit or units.



Perhaps most confusingly, AES42 also allows the microphone itself to manipulate its digital gain. To accommodate this, the DMC-842 can be configured to either use its own digital gain processing, or to send this information over AES42 and allow the microphone to perform this function (in this case the DMC-842 applies no digital gain, and the gain control acts as a remote for the microphone). In a mixed economy, it can also be configured to automatically detect the capabilities of individual microphones and use internal or AES42 gain accordingly.

In use, the unit performs admirably. The quality of the D-A conversion is fabulous — very open and full sounding with plenty of detail resolution. There's plenty of headroom on the analogue side too, and the outputs can be switched on the front panel so that 0dB FS equates to +13, +19 or +24dBu.

There are a few annoyances. Using AES42 to control gain at the microphone end has a very 'laggy' response — you twist the encoder and then wait half a second until the gain level switches. This is better when using the unit's internal digital gain controls, but still slightly apparent. And switching the gain setup between automatic and interface controlled with a mic plugged in resulted in maximum gain being applied for a brief period — ouch!

Sonically, the Schoeps CMD 2U and MK2S combo offered no surprises at all — character-wise it sounded just like a Schoeps. Detailed and neutral, with a gentle HF lift, it works beautifully at all but very close distances to sound sources. It's clear that Schoeps has taken quite a conservative approach to the 'digital microphone', which has advantages and disadvantages. There are the clear advantages of being able to use existing capsules and accessories from the Colette range, and there's a reassurance in a very familiar quality of sound coupled with the shortest signal path before A-D conversion. But this comes at the expense of not taking full advantage of the AES42 standard.

As far as the DMC-842 is concerned, it's clear that this is still something of a developing area in audio. RME seems to have come up with a product that, as much as is possible in a relatively new area, seems to be able to cover all the bases and cope with any AES42 implementation or digital interfacing requirement you can throw at it. Nice one. ■

PROS Great sounding D-A and SRC; flexible enough to cope with any combination of AES42 implementations; straightforward in operation; every sync option you could wish for; MADI option.

CONS AES42 remote control functions a little tortuous without the software remote control; gain control a little 'laggy' — especially over AES42.

Contact

RME, GERMANY:
Website: www.rme-audio.com
SCHOEPS, GERMANY:
Website: www.schoeps.de