THE ART OF DISCRETION

The recording venue—a recital room in Pfaffenhofen near Munich

On the mic

The AGM *MR-1* is a complete recording chain, beginning with a rather special 4-capsule microphone and delivering a 2-channel digital final output. Along the way it incorporates several fresh approaches and explores some established but under-used ideas. The system comprises the microphone itself, preamplification to line level, analogue-to-digital conversion, matrix processing of the four microphone signals, stereo reduction-encoding and an output section offering a variety of 2-channel formats.

The microphone itself is visually ▶

Dave Foister evaluates the startling 20-bit digital, discrete 4-channel *MR-1* microphone from AGM

Anyone who has been following the world of classical music recording lately will be aware we live in exciting times. This traditionally staid sector of the business has in recent years seen a explosion of development, with attempts being made to refine every stage of the recording chain and to find new solutions to old problems. This in turn has led to healthy heated debate, and there will be those who may find something to say about bandwagons and hype, but the general thrust has been positive and innovative. Players in the game cover a wide spectrum, and include small specialist manufacturers, such as Crookwood with the *Paintpot* mic preamp; influential names, like Yamaha, whose 20-bit mixers and recorders are in widespread use; and, of course, the record labels, notably Sony Classical and Deutsche Grammophon. These, and many others, have highlighted several key areas for re-evaluation, and attention has been particularly focused on:

1. The placing of microphone preamps locally to the microphones to minimise the length of low-level cable runs.
2. Remote control of those preamps.
3. Conversion to digits at the earliest possible stage in the chain.
4. Digital encoding using more than 16 bits.
5. Carrying out all processing and mixing in the digital domain.
6. Enhancing the sonic performance of the final 16-bit medium.
7. A move away from the big multimicrophone rigs and a reappraisal of simpler 'purist' microphone techniques.

What seems to have been missed by many in of the industry is the fact that since 1991 AGM Digital Arts have been quietly developing a microphone recording system which addresses every one of these key areas and does considerably more besides. I went to see it, and try it out for myself, during rehearsals for a chamber music recital at a beautiful intimate recital room in Pfaffenhofen, near Munich.

AGM Digital Arts' *MR-1*

very striking; it may appear bizarre to some but will be instantly familiar to anyone acquainted with the *Soundfield* microphone, with which it shares its fundamental approach. Like the *Soundfield*, it has four discrete microphone capsules, mounted in the form of a regular tetrahedron (the significance of this arrangement and the configurations derived from it are discussed in the sidebar 'A-format, B-format and Steering'). In the *MR-1*, the capsules are modified B&K *4011s*, carefully selected for matching and for off-axis performance; the electronics in the slim microphone stem are also B&K, as is the shockmount. The whole thing is much smaller than one might expect from the publicity shots, and is surprisingly elegant and unobtrusive.

The microphone is connected to its dedicated quadruple preamp by an 8-pin multiway terminated with screw-lock Tuchels, and although it is claimed that runs of up to 100m can be used without problems, the nature of the system is such that it is much more likely that this multicore will be kept as short as possible. The preamp powers the capsules and brings their four discrete signals up to line level to feed the convertors, which expect by default +18dBu for full modulation. The nominal gain is adjustable in 6dB switched steps from 0dB to +24dB, and each step employs a separate optimised amplification stage rather than varying the gain of a single stage. This should avoid the noise performance compromises of a single variable preamp.

The *MR-1* in operation over the stage

Although the gain is selectable from front-panel switches, the unit will normally be set for remote operation. In this mode, the main computer takes over control of the gain, allowing the rack containing the preamp to be sited right by the microphone. The remote switching is smooth and click-free, and the threat of control data breaking through to the audio is avoided by complete separation of the relevant boards in the unit.

In the same rack as the preamps sit the A–D convertors. These are AGM's own units, known as the *Dream AD-1 Classic*, and are built round Prism 20-bit convertors. Since they are the same devices as those used later for digital format processing, most of their features are not used at this stage, but a synchronised pair of *AD-1s* provide the two 20-bit AES-EBU outputs carrying the four signals from the microphone; this means that before it even leaves the bottom of the microphone stand the set of A-format signals can be encoded in robust digital form.

Given a suitable recorder—the Nagra *D*, with its four 20-bit-capable tracks, is ideal—these signals can be recorded directly on site and the full processing done elsewhere post-session, removing the problems of inadequate monitoring and difficult listening environments often encountered with location work. This, however, requires a considerable act of faith, since the A-format signals give no coherent idea of what the possibilities are for the final stereo image, or even whether the microphone is ideally positioned—although this is far less critical than with conventional microphone arrangements.

Where possible, then, the A-format signals will be fed to the main processors, with perhaps a ▶

A-FORMAT, B-FORMAT AND STEERING

The principles used to extract directional information from a microphone array such as that in the *MR-1* will be familiar to those who have worked with the *Soundfield* microphone but perhaps not to many others, although they are quite straightforward. The tetrahedral array allows the four capsule outputs, known as the A-format signals, to be combined and matrixed into a standard set of directional signals known as B-format, and this is a two-stage process. The capsules are arranged so that the first faces left, front and upwards (known as LF+), the second faces right, front and downwards (RF-), the third left, back and downwards (LB-) and the fourth capsule right, back and upwards (RB+).

Given that the capsules have a suitable polar pattern—somewhere around subcardioid—then subtracting the RB+ signal from the LF+ signal will produce a figure-eight pattern facing front-left in the horizontal plane (the vertical components will cancel out). Similarly, subtracting LB- from RF- gives a horizontal figure-eight facing right-front. Adding these two derived figure-eights gives a new figure-eight facing directly forwards, and this is known as the X component of the B-format set. In the same way, a figure-eight facing to the left can be derived, and this is known as the Y component. A similarly obtained upwards-pointing figure-eight is the Z component, and a straightforward sum of all the capsule outputs gives an omnidirectional response which becomes the missing W component.

The first thing to notice is that by gradually replacing X with Y in suitable proportions (sine-cosine proportions to be exact), and doing the same with Y and -X, the whole array effectively rotates about its vertical axis; this is what the Azimuth control of this type of microphone does. A similar crossfade between X and Z produces an up-down swing about the side-to-side axis, and this becomes the Elevation. Furthermore, simply inverting the phase of the Z and Y components inverts the whole microphone, and swapping the roles of X and Z switches between side-fire and end-fire.

It is also apparent that simply adding X (figure-eight) and W (omni) will produce a front-facing cardioid; adding X and Y produces a figure-eight facing 45° left, which can be turned into a cardioid by adding W. By extension, any type of first-order microphone (omni through to figure-eight via cardioid) facing in any direction can be precisely generated, and given sufficient controls and outputs, any number of such microphones can be produced. Simulating a conventional coincident pair, with fully variable polar pattern, angle, and orientation, is therefore an easy task, as is an M-S configuration, where M is a mix of the X and W components and S is simply the Y component. ■

Location control room showing the twin PC control stations

4-track backup recording running alongside for later reworking if required. In the full *MR-1* system the processors live in two rackmounted PC-compatible computers with remote screens, keyboards and mice. The first computer deals with control of the preamp gains, matrixing of the A-format signals into B-format (or Component Audio as AGM prefer to call it) and subsequent steering and configuration of the microphone, all controlled from a familiar *Windows* environment with real-time metering, calibrated control sliders and pop-up menus for the various options. Four meters show the incoming A-format signal levels while another four show outgoing B-format after the all-important matrixing and steering.

There are only four controls on the screen, and they all operate with the mouse in the same way, giving the choice of direct dragging, incrementing with the top and bottom buttons, faster movement by clicking in the bars, or instant reset to the default position by clicking the right-hand button. The first control is a digital level trim, giving a +10dB to -20dB variation (in 0.1dB steps) on the coarse-set preamp gains before the A-format metering. This is followed by the two B-format steering controls for the Azimuth and Elevation, both giving the full range of control to 180° in either direction. This means that no matter how the microphone is suspended in relation to the performers, it can be made to 'point' directly at them (without, obviously, any physical movement taking place). The remaining slider is for front-back Dominance, often thought of as a Zoom control since it approximates to the effect of moving the microphone towards or away from the performers so as to balance direct sound with room ambience to the desired degree.

The controls all operate smoothly and intuitively, with only an occasional zipper effect when making very fast alterations. There will never be a computer substitute for the continuous 360° azimuth controls on, for instance, the earlier *Soundfields* and the Audio Design Pan-Rotate, but the end result is the same and settings can be achieved quickly and precisely. Future software may feature rotary on-screen controls, but the sliders do the job well enough for me.

What they do not do is provide any way of directly synthesising a conventional microphone or stereo pair. Production of first-order microphones, pointing in any direction, is so straightforward from B-format (see sidebar) that it seems, perhaps, a curious omission, although eventually there may be a library of off-the-shelf configurations available, simulating traditional techniques and familiar microphone characteristics. This would be made possible by the file storage facility, which at present allows the current setup of the system to be saved for future recall.

Detail of the displays from the *MR-1*'s twin PC control stations

Pop-up options include various diagnostic setups, which can bypass various stages of the processing or produce a tone cycle, where all the B-format outputs generate 1kHz pulses in sequence. I would like to see more possibilities included here; I know from experience how hard it can be to trace a problem in a B-format system, ►

AMBISONICS AND UHJ

The most spectacular use of B-format is the generation of a complete Ambisonic surround sound stage, including height if required, matrixed on to as many loudspeakers as are appropriate for the listening venue. A feature of the resulting soundfield is its relative independence of listening position; it is even possible to locate sounds with some accuracy and to detect movement when listening from outside the ring of speakers. Where 4-track storage is not available for the discrete components, encoding to fewer channels (using a system called UHJ) gives very good results; since most recording and transmission media are 2-channel, this is the most commonly used version, although this obviously cannot encode height. Decoding this to multiple speakers gives results almost indistinguishable from raw B-format signals, with the exception that rear images are less precisely located.

It cannot be denied that Ambisonics and UHJ, following so closely behind the damaging quadraphonic debacle and suffering from confused direction and marketing, has had a pretty lukewarm reception down the years. People who have heard its full capabilities, however, have a tendency to become instant converts, and there is no doubt that UHJ encoding, far from being incompatible with coded playback, actually produces an extremely natural, powerful stereo soundstage with an unrivalled sense of depth and of the space in which the recording was made. A typical comment from listeners is that the loudspeakers seem to disappear.

Some of those who have espoused Ambisonics most enthusiastically have had their own preferred working methods and have produced results which, while pleasing to many, have not perhaps been to everyone's tastes. The unfortunate consequence is a body of people who believe that Ambisonics is only capable of producing a warm, ambient, reverberant recording, and has difficulty achieving immediacy and impact; this is decidedly not the case, as I would urge people to find out for themselves. ■

▶ where one misbehaving microphone capsule can cause strange effects on all the outputs. For these purposes mutes and-or solos on each of the A-format signals would be enormously useful.

The steered B-format signals pass to the second PC, again via two 20-bit AES-EBU links which could, if desired, be recorded onto 4-track for later stereo processing. Normally, however, these go straight to the software responsible for producing the final 2-channel output, which will be UHJ-encoded Ambisonic. This screen shows meters for incoming B-format (only three channels, since the vertical Z component is not required for UHJ and is therefore discarded) and outgoing 2-channel UHJ. There are three control sliders, one for level trim, one for stereo width and one for another front-back Dominance control, presumably on the assumption that B-format signals from another source, not equipped with such a facility, might be presented here.

This final output—still 20-bit—can be converted to analogue at this point for monitoring purposes. The setup I was using employed a Lake People D–A for this purpose, which has an auxiliary digital output to feed the final *AD-1* convertor. This is used in digital–digital mode, and can record direct to a 20-bit medium or via a choice of formats to 16-bit. Like most 20-bit systems, this needs to come down to 16 bits in the most civilised manner possible, and offers a choice between flat dither and Super Noise Shaping (SNS), which reshapes the noise out of the ear's sensitive midrange and into the HF band where it will be less troublesome. This is claimed to give a weighted signal-to-noise ratio of 110dB from 16 bits, and its effect when monitoring via a connected DAT machine is quite distinct, with a clearly perceptible increase in perceived transparency.

A further mode offered by the *AD-1* is DRE (Dynamic Range Enhancement) which is a double-ended encode-decode process designed to yield virtually 20-bit performance from a 16-bit medium; the idea is that DAT could be used for temporary storage of the system's output prior to subsequent 20-bit editing or other processing. I was not able to try this technique, but it sounds like something we should know more about as a freestanding process.

Future plans include systems for combining multiple *MR-1*s into one B-format soundfield, ideas which have their germ in the digital mixer modules already available from AGM. These allow external mono spot microphones or other feeds to be added to the *MR-1*'s image with appropriate positioning and compensatory delays. The use of these is, in fact, vital since the processing involved in the *MR-1* system itself adds several tens of milliseconds of delay to the microphone signal.

With these and other options—including full Ambisonic decoding for up to eight ▶

Preamp and A–D convertor rack

▶ loudspeakers and UHJ to B-format conversion—available and in the pipeline. The *MR-1* clearly has the potential to operate effectively as a digital B-format workstation.

Conclusion

Despite the apparent complexity of the *MR-1* system, it is in fact extremely simple to set up and operate, and any effort involved (principally in lugging it all about) is repaid in the results it produces. It takes a special piece of audio kit to make the little hairs on the back of my neck stand up these days, but the *MR-1* managed it, with its startlingly natural, clean, quiet, accurate portrayal of the musical performance I had already heard going on in the next room. Its ability to convey the character of the room was uncanny, to the extent that in when we placed it in one particular position I thought something had gone drastically wrong, as the image became indistinct and positions hard to identify; when I checked in the room, it turned out that the microphone must have been at the meeting of several room resonances—the image live in the room was just as blurred at that point. Having established that, a more suitable position produced the now familiar transparent reproduction of the impressions received in the room itself.

It is important to be clear about the fact that none of this is 'vapourware' or lashed-up prototypes; this is a complete finished product, ready to go, and it works superbly well. This is a dream of a system, built with no compromises and with a careful selection of leading-edge techniques, which in the right hands will be an enormously powerful tool, justifying every penny of its price. There is no substitute for recording acoustic music, simply and to the highest available sonic quality, in an appropriate venue, and the *MR-1* has to be the most elegant, musical way of going about it. ■

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