AUDIO FOR BROADCAST, POST, RECORDING AND MULTIMEDIA PRODUCTION

resolution

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Mixing for radio, TV and Internet

Dutch mixer GIJS FRIESEN shares his experiences of working across a wide range of different programme and output.

As a freelance audio engineer/sound designer it happens quite frequently that I work for a postproduction company in the morning to do sound design/mix for TV commercials and then go to one of the radio stations I work for in the afternoon to mix a full band that is performing live on-air. They are two completely different worlds but until recently there was always one thing they had in common — the audio would go through some form of broadcast processing applied by the radio or TV station. This is still the case when it comes to radio but the way TV processing is applied changed a lot with the introduction of EBU R128. And, of course, there is now more and more material that has to be mixed for Internet only, where no processing is applied at all, except maybe loudness normalisation.

So how do you make a mix that sounds great on radio or TV while taking the processing into account? And how is this different from mixing audio for web material? In this article I'd like to look at this, share my findings and talk about what's going on in those 'magic processing boxes' that are used at radio and TV stations.

Before talking about broadcast processing, I'd like to discuss something that I think is at the basis of every good mix and that's a nice smooth workflow with tools you know well. In my opinion, there are three 'rules' that apply almost everywhere: Pro Tools is used 95% of the time in a studio where audio is recorded, edited or mixed; there is never enough time — before you know it the time is up and the next client is waiting for you impatiently; and even though there is never enough time, people still expect you to make the best mix ever made by anyone on earth.

In my opinion the best way to deal with this reality is to get your workflow right and learn to work quickly and efficiently with the tools you have/like. This will greatly speed up your working process so you can focus on what really matters — and that's making a great mix.

I work in audio postproduction and radio and because of the popularity of Pro Tools in postproduction, it is a no-brainer for me to work with it. Besides that I like Pro Tools and I use my own Pro Tools template (stored in my Dropbox), with all the tracks routed to the right buses and some plug-ins where I like them. In my template I only use Waves plug-ins and that's not because I like them so much (even though I think they sound more than decent), but because every post studio I have ever worked for seems to have at least the Waves Gold Bundle. This gives me the benefit that I can start working with my template virtually anywhere in no time without ever having worked there before. I open my template, play a reference track to find out what the monitors/room sound like, and off I go. This is expected of a freelancer these days.

Over time I usually start using other plug-ins or change my workflow depending on the studio I'm working in but my template gives me a starting point. I also use Pro Tools in my own studio with that same basic template and it happens sometimes that I start a project somewhere else and get asked to finish it in my own studio.

Mixing music for live radio obviously needs a different way of working and usually I use a digital live mixer, like a Soundcraft VI4 or Yamaha LS9, most of the time with outboard — like the Neve 33609 compressor, which is very popular at radio stations here in The Netherlands and is a great sounding compressor. Instead of analogue outboard you can use plug-ins and the VI4 has a great tool for this in the Realtime Rack. With this you are able to use UAD-powered plug-ins (controlled via touchscreen) on channels/buses of the VI4.

There is never much time for a sound check so to speed up the mixing I use 'standard' settings on channels, which is a common thing for mixing music for live radio. For example, my snare drum channel has a high pass filter on it, an EQ with a dip around 400Hz and small boost around 2-4kHz, and whatever radio station I work at I tend to use these settings as a starting point. Sometimes there is no time for a sound check at all, so my mix has to sound at least OK as soon as I open the faders and balance them. All the fine-tuning is done during the live performance in these cases and it is important to know the sound of every microphone involved so you can predict a little what things might sound like even though you haven't actually heard anything yet.

Then it's time to focus on the mix and if you're mixing for radio or TV start to worry about the broadcast processing. The first question is: why do we need broadcast processing at all?

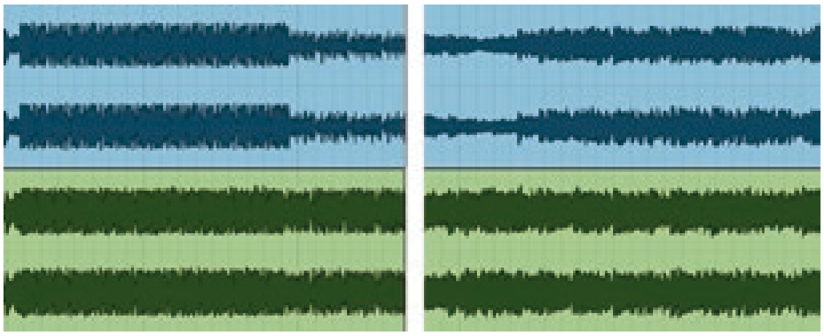
Gijs mixing music in the Radio 538 Live studio in Hilversum, The Netherlands.

When broadcasting radio, some form of limiting is usually needed to meet the technical requirements of, for example, good old FM. Besides that you may want to apply some form of levelling to make sure the audio has more or less the same sort of loudness over time. But lots more processing is applied than just this — radio processing is used mainly to create a 'unique sound identity' for a radio station and, of course, to sound just a little louder and clearer than any competing radio station. No, the loudness war isn't over in the world of radio... at least not here in The Netherlands.

When it comes to TV things are a little different. Some stations used to use very heavy processing for the same reasons as radio stations — to colour the sound in a way that stands out from the rest and gives your station a unique 'sound identity'. With the introduction of R128, nowadays there are stations that use next to no processing at all (except for some limiting perhaps to prevent overshoots). When mixing for TV I think it is not processing you have to worry about anymore so much as the small (and cheap) speakers that are used on most TV sets. The same goes for mixing for Internet productions as they are played through small speakers in iPhones, iPads or laptops. We will look at this later.

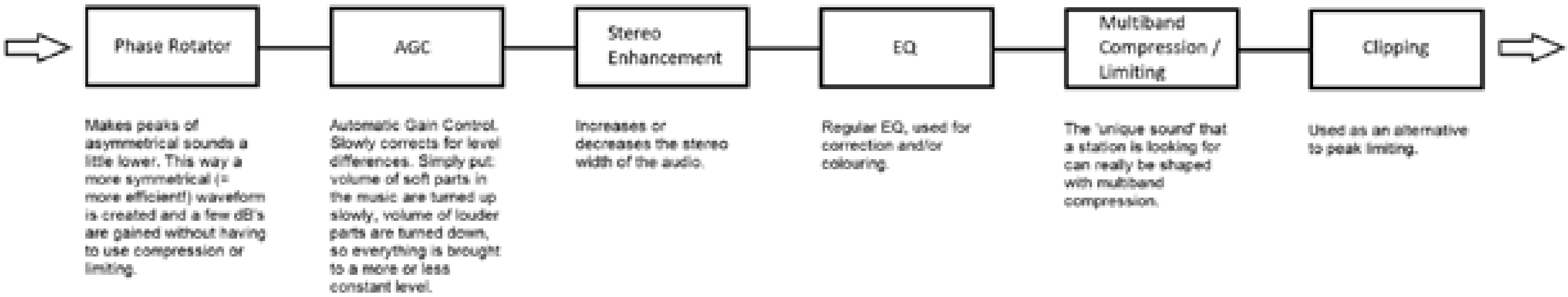
So what's going on inside those 'magic boxes'? It would take too long to go into every technical detail but a brief overview of what is going on can really help to understand why some things work or don't work on-air, especially on radio. If you are interested in a more detailed explanation of what's going on in broadcast processors, I suggest the part in Bob Katz's Mastering Audio on 'Radio Ready: The Truth'.

Many radio/TV stations use an Orban Optimod and there are various types for radio and TV. These multiprocessing units take care of everything the station would want processing-wise, from multiband compression to stereo widening to EQing to limiting. An Optimod contains a chain of processors. Stations use some, or sometimes all, of the available processors to create 'their' sound. See figure for an overview of the processing chain. The order in which the processors are applied varies but the order as shown in this figure is pretty common.



Audio before and after it has gone through an Orban Optimod. The blue waveforms represent the original masters. The green waveforms are the same masters played through an Optimod. You can see that the dynamics are gone and the soft parts of the songs are just as loud as everything else.

You make a great mix for Optimods by not trying to work around the processing. For example, the high end may sound exaggerated on the radio but mixing with less high frequency to correct for this simply doesn't work. All the songs are played through the same processor with the same amount of processing and your 'corrected' mix will just sound dull compared to everything



else. If your mix sounds good before processing, it will still sound good after processing. Extremely flat and with a completely different tonal character perhaps, but still well balanced.

Another myth: 'if I add more compression and limiting to my track and maybe turn down the master volume a few dBs, it will easily "slide" through the broadcast processor without its compressors working on my track.' Unfortunately it doesn't work this way. One of the reasons for this is that one of the first stages of every broadcast processor is the AGC, as described before. This will correct for level differences in your mix, so it enters the rest of the processing chain at a more or less constant level. After that it will reach the compressors and clippers and it gets processed just as much as all other tracks.

Talking about clipping, it is not a good idea to use clipping as part of your mastering process if it is just used to make your song sound louder on the radio. It actually works counterproductively. The more clipping you apply at the mastering stage, the less clear, punchy and defined your mix will sound on-air; definitely not louder.

I am not saying compression or limiting are bad and let's not get scared of using them as a reaction to the loudness war. They are very useful to even out a vocal track, to make a snare a little snappier, or just wherever it sounds good. In postproduction a healthy amount of compression and/or limiting on the voice-over track is common. Together with the right EQ this makes the voice-over sound big and up-front in your mix. If you don't apply this processing, you will obviously hear the difference on the radio.

But the difference in loudness between a compressed/limited mix and a heavily compressed/limited and clipped mix will not be heard on the radio. It can be heard in terms of sound quality though: the clipped mix will just not sound as good/clear. The same goes for too much limiting.

In my opinion there is one very important thing to aim at when mixing for heavy broadcast processing and that is mixing with space and depth — make a '3D' mix. Choose what instrument is mixed up-front and which instruments are mixed a little more to the back, using mic placement (close/distant miking), EQ (e.g. more presence to make the instrument sound more up-front) and reverb. This is, of course, nothing new as creating space and depth has always been a crucial element of mixing but when mixing for radio (where heavy processing is applied) it is even more important. When every instrument has to fight for its place in the mix, some instruments will just get lost when played on the radio and your mix will start sounding very messy.

I think 3D mixing is also important when mixing for TV — or mixing web videos — to get the mix to sound right on small TV/laptop speakers. This will certainly help a lot.

Another important thing when mixing for broadcast processing is creating a clean mix. Filter out as many unwanted frequencies as possible. Let the bass guitar take care of the low end and filter off the lows of the guitars for example. This can prevent nasty phase problems that may occur and cause all sorts of trouble, again especially on radio.

When mixing for TV (or the web) also be careful with frequencies below say 75Hz as they may not get through the average TV or laptop speaker. A heartbeat sound for example can be problematic. It may sound nice on your studio monitors but it will almost completely disappear on the speaker of your laptop/TV. Checking your mix on an actual TV or via the speakers of your laptop helps. You will probably also notice that the level of the music track compared to the voice-over may sound as if it is too loud on big speakers, but sound just right on your TV/laptop. Since processing is used less and less these days on TV, mixing through an actual TV gives an excellent indication of how things will sound when broadcast.

A good mix will sound good anywhere. Although mixing TV commercials or web videos is completely different from mixing music in a studio, the same basic principles still apply. A good mix starts with the right workflow and besides that I think that in all cases getting a sense of 3D and space is needed to make your mix sound good wherever it is played, especially when heavy broadcast processing is applied. What is crucial is the right tonal and dynamic balance, which can only be found by experience.

Even though there are big differences when you compare audio post with mixing music for live radio, there are also a lot of similarities. Once you learn to deal with the differences it can be a big benefit to you as a freelancer to specialise in more than one aspect of the audio industry because you're likely to get more work. For me, the most important thing is that I learn things when mixing music and use that experience when sound designing and mixing a web video or radio/TV commercial. It works the other way round too and you end up getting better results, which is good for everybody.

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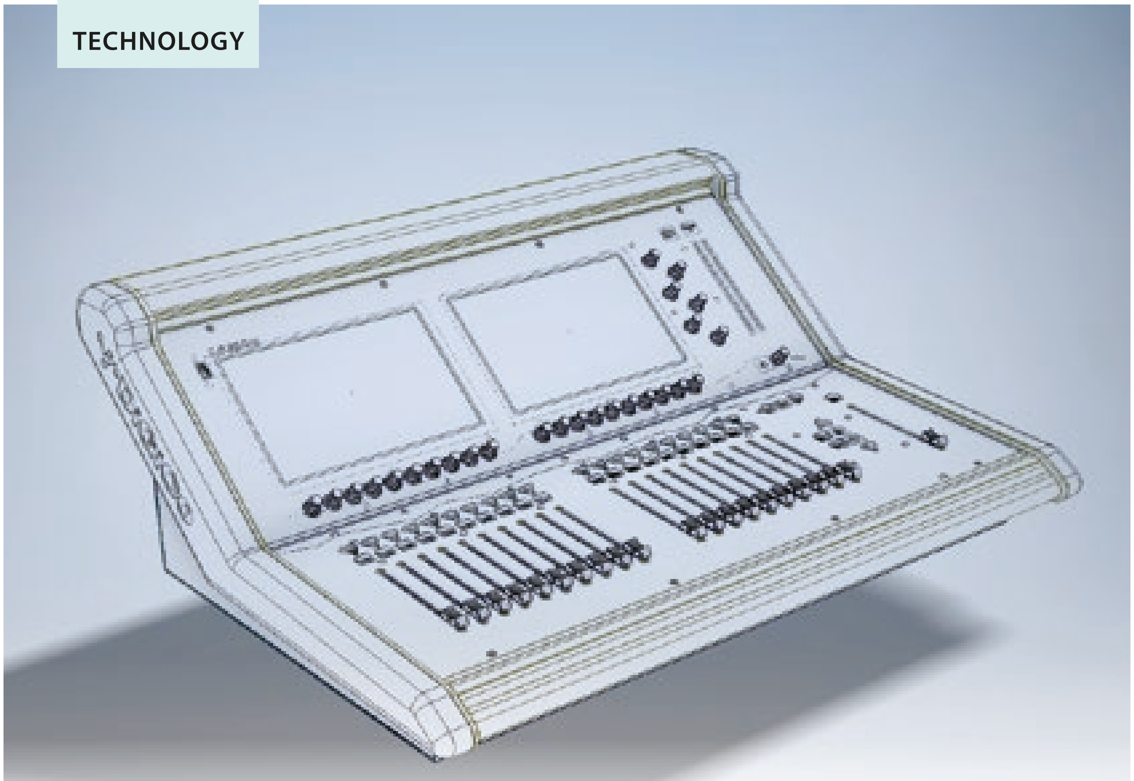
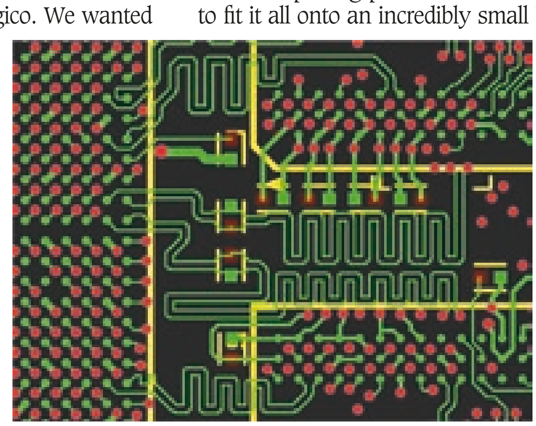


Developing S21

Digico's S21 console is the most affordable in the manufacturer's range. Digico's MD James Gordon and technical director John Stadius explain how they did it. ZENON SCHOEPE

'S21 is a console we have always wanted to make,' says Digico MD James Gordon. 'But we didn't want to do what most manufacturers do and simply make a lower cost product and brand it Digico. We wanted to make a proper Digico console that we could offer for a lower price point than our existing SD series consoles. We couldn't do that until John Stadius and his R&D team could develop a technological advantage that would deliver the same audio quality and operator experience, combined with the visual feedback and all the other attributes that Digico is famous for.' The S21 story started three years ago when Stadius and his team came up with the idea of moving away from the embedded PC structure that the SD series console design is based around, to a lower cost solution. It meant reinventing how they thought about things and starting again to create a new product with a design shift that is as radical as the SD7's move to FPGA technology was. 'With S21, we've moved from a traditional Intel embedded PC to a different type of processor core altogether. This has meant a complete rewrite of the operating system — we use Linux on S21 — and are running a completely different application code as well, which has involved rewriting all our user software,' says James.



'We are still using FPGA technology — the key that unlocks the flexibility that our R&D team needs to specify and deliver Digico consoles — in S21, but it has moved on an awful lot since we first used it. Crucially, the components have got wider in terms of choice and for S21 we were able to find a lower cost chip that would allow us to port across the audio quality of the SD Series to this new console; we use exactly the same algorithms, just running on a smaller device. And it's 96k as standard — we feel strongly that 96k is now the only way a professional console should be.'

'We're using Quad Core ARM processors in the work surface of S21 to make the code more "common",' adds Digico Technical director John Stadius. 'ARMs are in everything these days — your phone, your iPad, your washing machine. They have what we call a RISC (Reduced Instruction Set Computer) processor, which is a very simple core, running very small instructions, at a very high rate. It's very efficient, very low cost to manufacture, and very flexible.

'In fact, it's not just a processor,' he continues. 'We call this a system on a chip, which includes the Quad Core ARM, four graphics controllers, Ethernet controllers, USB, etc. — basically you have a whole computer on one piece of silicon. It's very powerful. All you have to do is put some memory and all processing on the outside and hey presto! You have a mixing console.'

This is, of course, one of John's deceptively simple explanations as the development of S21 was a considerable challenge. 'The FPGA part was the easy bit, because we've used them for five or six years and we're getting more and more from the devices,' James continues. 'As existing SD customers know we've added new features and extra processing, as we've become more efficient with how we control and master what gets dropped onto the FPGA.

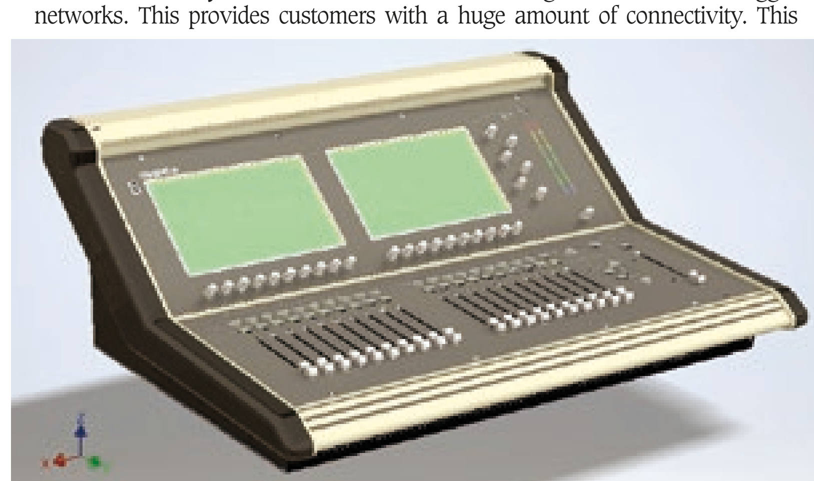
'With S21, we decided to make things simpler, both in terms of what's going on inside the console and from an operator perspective. SD consoles have gained a huge user base because they're so flexible. You can do pretty much anything on them and most things in more than one way. Audio engineers absolutely love them because there are really very few restrictions. The trouble with that is that you have to really know what you're doing, or you can dig yourself into a hole and not necessarily know how you got there, especially on something like the SD7, where you've got hundreds of channels and buses all with full processing.

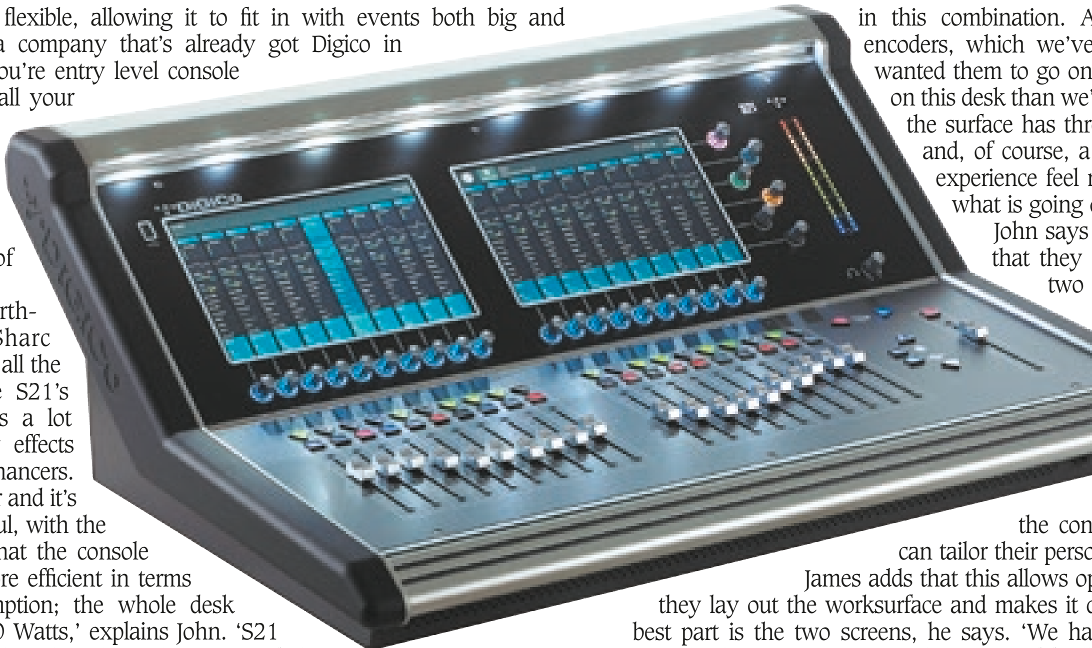
'We wanted to make S21's interface easier to use for a larger audience. So we redesigned the graphics, but still kept the Digico feel — when you walk up to S21 you can see that it's got all the channel strips that you have on an SD console. We've taken the colour indication from the SD7, as well, and used its Hidden Til Lit technology, which is a really nice way of tying rotaries to what they're actually controlling on the screen. And we've refreshed all of that with a more 'flat' graphical design, so it's a bit like an iPad or a modern app, which makes the whole thing very user friendly.'

On the worksurface, Stadius was determined that there would be two large, multitouch screens. As a result, keeping costs down has been a real drive on procurement. Under the hood, the team has included elements from its accessory business, porting products such as UB MADI across to S21, but still managing to fit it all onto an incredibly small PCB.

'Another very smart addition is S21's DMI card slots,' says James. 'One of the things we've always done very well with our Racks is the modularity of being able to have different formats. But now, it's increasingly common to have a lot of different formats and this requirement is only going to increase. With S21 you can plug in these different formats simply by buying different DMI cards for the slots, allowing customers to mix and match.

'We also do 16 line input and 16 line output cards which means you can add basic I-O if, for example, you need more analogue ins and outs for a small corporate show. Alternatively you can plug a SoundGrid or Optocore module, or even a Dante or Calrec Hydra 2 interface, and connect straight into one of our bigger networks. This provides customers with a huge amount of connectivity. This





makes S21 very flexible, allowing it to fit in with events both big and small. If you're a company that's already got Digico in your inventory, you're entry level console can connect into all your bigger consoles; if you're a smaller company, you can now get to experience the sonic quality of Digico.'

A new, fourth-generation Sharc processor controls all the algorithms in the S21's FPGA, as well as a lot of the non-linear effects such as audio enhancers. 'It runs a lot cooler and it's a lot more powerful, with the knock on effect that the console is enormously more efficient in terms of power consumption; the whole desk consumes only 50 Watts,' explains John. 'S21 uses a ten-layer board and employs one of the best memory layouts in the world. The speed of the logic on this, compared to the old days, is phenomenal. We're running signals up to 3Gbits. Also, the interconnectivity of these chips — which are getting more and more dense, with less than a 0.5mm pitch on the pins — is more serial, so you use just a pair of tracks, rather than the eight or 16 you needed when they were in parallel. The trouble with the parallel set up was that as the speeds went up, you had to make sure that the tracks were exactly the same length, which is impossible when you get up to 64 tracks. Now, rather than sending 64 tracks at, say 200MHz, we're sending just two at 3GHz. That makes the layers of the board a lot simpler.

'The board self-time-aligns, catching the packets and realigning them using a synchronising clock,' he continues. 'This means that the tools we have to use to design the boards are a lot more complicated; you can't just lay the tracks and manufacture them, you have to simulate them to make sure they're going to work beforehand.

'What we do now — and have been doing since SD7 — is to use a bit of software that does a continuity check on the board before you send it off to be made. It also does a signal integrity test, making sure the signals are the shape they should be — nice and square with no bouncing or ringing — and a thermal analysis, to make sure no parts are getting too hot. This all goes on as you go through the development process, which makes it much quicker and gives you more confidence that what you're doing is going to work.'

Most serious high tech companies do board layouts in this manner according to James, but not so many in our industry. 'What's unique, from a Digico perspective, is how we've applied Sharcs, an FPGA and an ARM to deliver at a very low price point. At the moment, no one else in the industry does that in this combination. Also, we have touch-sensitive encoders, which we've never done before, but John wanted them to go on because we have less controls on this desk than we've had before. Each encoder on the surface has three functions, a touch, a switch and, of course, a turn. This makes the operator experience feel right and it's very fast to sense what is going on and to just mix.'

John says that another unique element is that they have included multitouch over two screens — 'You don't tend to find that anywhere,' he says. 'We've integrated ours so you can touch and drag from one screen to another. This is a real advantage in getting channels from one side of the console to the other, so operators can tailor their personalised console's layout.'

James adds that this allows operators to be flexible with how they lay out the worksurface and makes it quick to change things; but the best part is the two screens, he says. 'We have 15-inch screens on the SD consoles because we want to have lots of feedback. On the SD9, for example, there's a huge amount of feedback with just one touchscreen. On S21, we have smaller screens, which meant we couldn't get enough feedback on just one. Two achieves speed of operation and the security of knowing what's going on.'

The S21 opens up new opportunities for Digico in the broadcast and studio. 'There's a big requirement for small location mixing consoles, which wasn't really covered by the SD series,' says James. 'The fact that S21 can link into the bigger Digico's, and can also link into Calrec consoles, will open things right up. The fact that you can put a Hydra 2 card on the back of the S21 means that if you are working on a broadcast where the main mix to air is being done on a Calrec, you can plug a Hydra 2 card into the back of your S21 and have 64 connections straight in and out of Hydra and share their I-O, or they can share yours.'

Unlike the SD series, there are no application-specific options for the S21 because of its price point but it does have the flexibility to be applied to different scenarios and the connectivity to other parts of the broadcast environment to make it a very useful product. 'In the studio market, the DigiGrid brand has become quite prominent, and having a console that can connect into that via SoundGrid will allow S21 to sit on a SoundGrid network as well,' says James. 'We're already getting a lot of interest from smaller studios. And because there's more horsepower that can be unlocked, as with all our products, we have the ability to evolve S21 over the years.'

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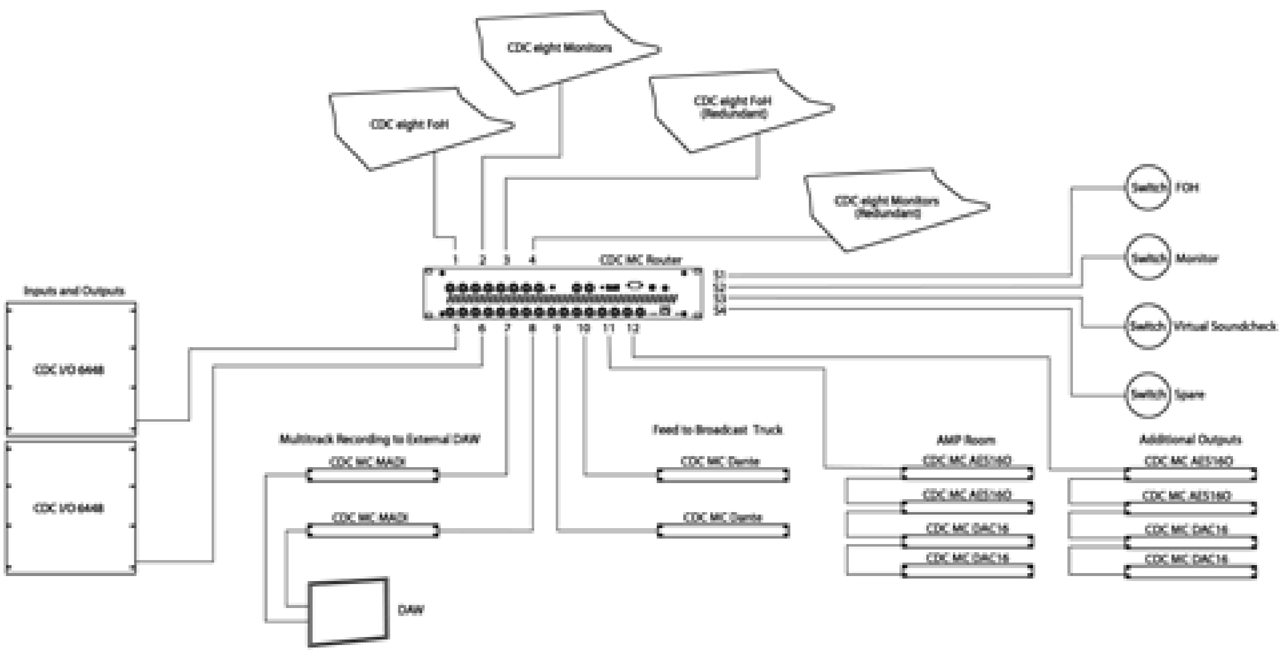
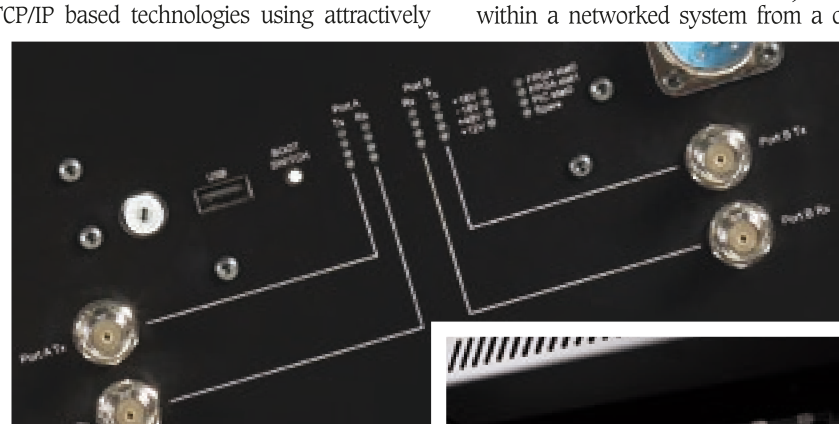


This example shows 128 analogue inputs shared between four consoles, which can be configured either as four discrete mixing systems or as two redundant console pairs. The diagram also shows audio distributed to remote broadcast and recording facilities via MADI and Dante conversion. The Cadac CDC MC Router provides gain compensation for all audio streams should any analogue input gains be adjusted during the performance, and it can be preprogrammed with up to eight different routing maps, which can be selected from a front panel rotary switch or remote switching panel. All cables can be up to 150m long.

Cadac MegaComms explained

Cadac has developed its own digital audio networking solution. Cadac's RICHARD FERRIDAY presents the company's beliefs on the available technology options and discusses how the MegaComms system works.

When Cadac was researching a digital snake solution for the S-digital console in 2005, there were a number of accepted and well established ways of transmitting multichannel digital audio from a stage to the mix position. Some were well established methods, such as MADI, and other new TCP/IP based technologies using attractively low-cost computer industry hardware. Unfortunately, none of the existing systems met the Cadac criteria for extremely low latency, high channel count and high sample rates. The Ethernet-based protocols were found to be particularly unsuitable for live sound applications due to their long and unpredictable propagation delay. In an environment where more and more audio processing and transmission is performed in the digital realm, accumulated delay becomes a significant problem. With digital wireless microphones, consoles, plug-in processing options and digital in-ear monitor systems all contributing about 2ms delay each; the accumulative latency can cause difficulty for many performers to keep in time and in tune. This issue is exacerbated when the Ethernet error-correction system is heavily engaged, and audio samples are processed with varying latencies causing time-smear and resulting in highly undesirable audio artefacts such as comb-filtering. Cadac was determined to maintain phase coherence in its digital consoles, something which is critical to achieve the audio performance for which Cadac is renowned. This requirement ruled out the use of an Ethernet-based technology, where the lack of deterministic latency means that absolute phase coherency is unachievable at latencies that are acceptable in live sound reinforcement.



Cadac was therefore obliged to develop a bespoke proprietary system, as a dedicated and specialised method of transporting multichannel digital audio within a live sound environment. The solution for this was the development of Cadac MegaComms technology, which has now expanded into a range of products as well as being adopted for the Cadac CDC eight, CDC four:m and now the new CDC six digital consoles.

At first glance, MegaComms appears similar to MADI. It uses separate send and return cables, BNC connectors and coaxial cable. However, MegaComms is a robust TDM (time division multiplex) system and means that propagation delay is reduced to a single tick of the system's 96kHz clock, reducing propagation delay by several orders of magnitude when compared to that present in most of Ethernet-based audio networking. The TDM format also means that each audio sample is locked to that clock, so system latency is deterministic (fixed at a single value) and not allowed to drift depending upon local environmental conditions, such as electromagnetic interference or increased network error-correction activity.

Control data is embedded within the data stream so no audio channels have to be sacrificed for this purpose. The high bandwidth available means that the current implementation of MegaComms can carry 128 channels of 24-bit, 96kHz audio plus control data, plus clock, bi-directionally, up to 150m via a pair of RG6 coaxial cables.

In addition to audio and control data, MegaComms provides for accurate, phase-aligned clock distribution, by embedding timing markers in the data stream. This allows reliable, low jitter synchronisation of all hardware elements within a networked system from a common clock source, without having to run any additional cabling or any additional setup procedures.

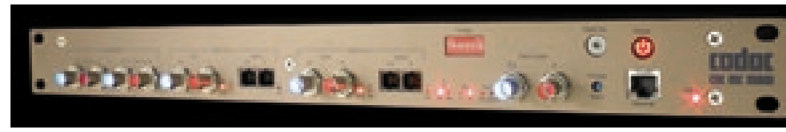
Most Cadac CDC digital consoles and I-O devices feature two pairs of MegaComms ports and the function of the second pair of ports is to provide the option of connecting a pair of redundant send/return cables. The redundant connections are active, so the device will automatically route audio, data and clock via the secondary connection in the event of a cable failure.

The simplest implementation of a MegaComms network is a straight forward console to stagebox configuration. In this application, the console provides the clock and the stagebox synchronises itself once the connections are made. Total through-system propagation delay for this system, including all console processing and A-D/D-A conversions, is 37 samples (@96kHz), or just under 385µs. This compares with the many millisecond propagation delays usually found in most other similar systems. For larger, true audio networks, a MegaComms router is required that provides low jitter clock synchronisation and flexible routing capability for up to 3072 channels of 24-bit, 96kHz. The addition of a router into a system adds only an additional sample (approximately 10µs) of propagation delay. MegaComms is also capable of providing automatic fail-over to redundant spare connections. Most Cadac MegaComms devices (consoles, large-format stageboxes and network bridges) support dual redundant connections with automatic seamless change over in the event of cable failure.

The hardware 'hub' for a large-scale network is the Cadac MegaComms router — the CDC MC Router. This device has 12 pairs (send and return) of MegaComms ports, easily identifiable from glowing colour coded BNC sockets. The MegaComms router can connect up to 12 MegaComms devices, and a MegaComms device can be a console, stagebox or network bridge. The format of the network is that of a star-type. Depending upon the programming of the router this can be a single or dual-redundant star configuration. Routing maps (up to eight presets) are installed into the router from a laptop computer via a standard RJ45 network port. Any of the eight map presets can be selected either from a front panel mounted rotary switch or from a dedicated hardware remote control.

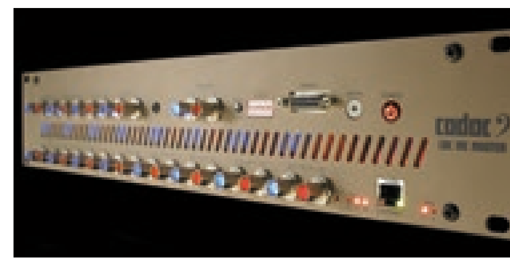
The CDC MC Router also has the useful function of providing 'gain compensation' in that if any of the connected mic amps have their analogue gain adjusted, the router will automatically compensate for the change in audio level to any and all other devices connected. This gain compensation process takes just a single sample (about 10µs) for the router to make the necessary adjustment, and so is completely inaudible in operation. This facility allows a stagebox's analogue inputs to be shared between a number of consoles, and for multiple consoles to be connected to a common set of I-O to provide multiple mixing systems or multiple fully dual-redundant systems. In these large scale applications, the MegaComms router provides the clock for all consoles and I-O devices.

The disadvantage of designing a bespoke, specialised proprietary system is that eventually, you may need to connect to another type of network for distribution purposes. For this reason Cadac has designed network bridging devices to enable MegaComms to connect to other protocols. All MegaComms bridges feature dual-redundant power supplies and dual-redundant connections to Cadac CDC consoles and Cadac MegaComms routers, as well as asynchronous sample-rate conversion to other popular protocols. MegaComms networks are clocked at 96kHz, but Cadac does provide for sample rate conversion to other clock speeds and conversion to more widely adopted protocols such as MADI and Dante.



CDC MC MADI — This converts 128 (64 x 64) channels of MegaComms 96kHz audio to 128 (64 x 64) channels of MADI @48kHz or 64 (32 x 32) channels at 96kHz. It features coaxial and optical MADI connections, dual-redundant MegaComms ports, dual-redundant integral power supplies and separate Word clock input for use with a standalone clock. It also supports connections to dual-redundant MADI streams.

CDC MC Dante — This converts 128 (64 x 64) channels of MegaComms 96kHz audio to 128 (64 x 64) channels of Dante @48 kHz or 64 (32 x 32) channels at 96kHz. It also features dual-redundant MegaComms ports, dual-redundant integral power supplies and separate Word clock input for use with a standalone clock. It also supports dual-redundant Dante networks.



CDC MC Router — This has 12 pairs of MegaComms ports, supporting up to 3072 audio channels and dual redundant integral power supplies. It has fully programmable routing via a PC network port and routing map selection (eight presets) is via a front panel switch or optional hardware remote control. Plus separate Word clock input for use with a standalone clock.

Contact

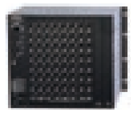
CADAC, UK
Website: www.cadac-sound.com

Products

Recent console introductions.

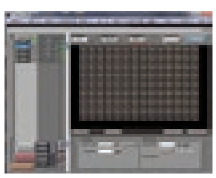
Digico

Designed to support and expand Digico's SD range higher sample rate I-O solutions, the D2-Rack comes with BNC or Cat5 MADI connections, allowing it to be used with a number of Digico consoles or as a standalone unit.



By using the latest convertors, the D2-Rack offers a more compact, more efficient, more affordable rack solution for connection at 48kHz or 96kHz with no I-O reduction.

The D2-Rack offers two I-O versions the first having 48 mic, 16 Line outputs, and two blank output slots allowing an additional 16 outputs in the owner's desired format analogue, AES and Aviom. The other version offers 24 mic, 24 AES inputs, 16 Line outputs, and two blank output slots allowing an additional 16 outputs in the owner's desired format analogue, AES and Aviom.



V685 software for Digico consoles includes an increased bus count for the SD9 from 16 to 24 Flexi Buses, SD11i/B input channel count increased from 32 to 40 Flexi Channels, support for Optocore DD4MR, DD2FR, X6R and DD32R devices in audio I-O, any SD5, SD8, SD9, SD10 and SD11 running Waves 9.5 will now have 32 stereo Waves racks, the availability of Aux Sends on Groups in the live and theatre versions, and support for the D-Rack AES input card and the addition of the D2 Rack as an I-O device.

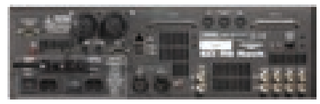
For the theatre, the Relative Faders in Cue Groups is now a Macro command, Auxes, Groups and Matrix channels can now be added to channel sets, and Channel Cues now defaults to showing names. For broadcast applications, Backstop PFL is now functional on output buses and there is a new option on the SD7B and SD10B for Speaker Mute to Dim.

A new version of Optocore for all optically-enabled Digico consoles provides connectivity to X6R, DD2FR and DD4MR units, allowing Optocore devices to live on Digico's optical loop, allowing X6R and V3R mic preamps to be controlled directly from the console. Users can add an Optocore 16-channel X6R-FX interface to the Digico network to provide additional I-O connectivity together with ethernet and RS485/422. They can also use Optocore DD2FR-FX and DD4MR-FX MADI interfaces to increase the number of MADI ports available on the console.

Digico is the only console manufacturer to use OEM Optocore and runs the native 2.21 Optocore protocol.

Digico's affordable S21 console includes aluminium extrusions, RGB switch encoders with HTL (Hidden Til Lit), Polycarbonate overlays and two large multitouch screens. It employs the FPGA algorithms of the SD7 and the mic preamp design is lifted from the 192kHz SD-Rack. It is 96kHz as standard with 24 mic/line inputs, 12 analogue outs, two AES I-O, Word clock I-O, one GPI and one GPO, DVI out (for an external monitor), two DMI slots (up to 64 I-O slot), two Ethernet connections for networking, 21 touchsensitive moving faders, four layers of banks of 10, Snapshots, 46 buses, 16 assignable 32-band graphic EQs, eight FX engines, four assignable DigiTubes, four assignable multiband compressors, one compressor and one gate per channel and bus and user definable Macros.

The Redundant Engine and Fader Pod provides a 'RE-assuring' partner for the SD10 console. The SD10 Redundant Engine (SD RE) is an engine in a box that provides connection to a monitor and a keyboard/trackball for SD7-style dual engine redundancy. It can be connected to a 12-fader remote worksurface, screen, keyboard and mouse and duplicates the functionality of the SD10's centre section.



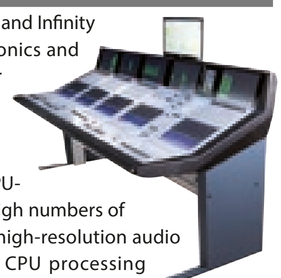
By connecting the box to the console with an Ethernet crossover cable and the system's audio racks, using MADI or Optocore, the SD RE provides a seamless backup for the console.

For broadcast, the SD5B's worksurface is a low noise, heat dissipating worksurface benefiting from Hidden-til-lit (HTL) technology, and its five digitally-driven full colour TFT LCD screens, three of which are touch sensitive, have a new configuration that allows easy access to single or multiple users.

There are also two interactive dynamic metering displays (IDM) and quick access buttons are positioned down the left side of the channel screens for fast and easy navigation.

www.digico.org

Studer

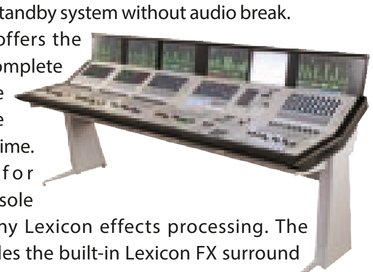


Studer's Vista X digital console and Infinity Processing Engine offers Vistonics and FaderGlow on the interface for control of 800 or more audio DSP channels and more than 5,000 inputs and outputs.

The Infinity DSP core uses CPU-based processors to provide high numbers of DSP channels for large-scale, high-resolution audio processing and mixing. The CPU processing provides a scalable system, with faster development of new signal processing designs, high channel counts, full system redundancy and the possibility of running third-party algorithms.

The Infinity DSP engine provides 12 A-Link high-capacity fibre interfaces for more than 5,000 inputs and outputs. A newly designed high-density I-O system — D23m — is used to break out these A-Link connections to standard analogue, digital and video interfaces. The A-Link interface also provides direct connection to the Riedel MediorNet distributed router, allowing many Infinity systems to be connected together with router capacities of 10,000 square or more.

A key element in the design of the Infinity Series is the avoidance of a single fault causing loss of audio and the Vista X features four processors, offering complete redundancy of the control surface. Now, the Infinity Core, with a combination of CPU-based DSP and A-Link audio interfaces, offers N+1 redundancy of the DSP engine and I-O with instant switchover between main and standby system without audio break.



The design also offers the possibility of two complete Vista X surfaces to be working on the same project at the same time.

Version 5.1 for Studer's Vista 1 console adds sister company Lexicon effects processing. The new software enables the built-in Lexicon FX surround processors — 2 engines with 8 mono effects processors — to be controlled directly from Vistonics.

The 52-fader Vista V is based on the same Quad Star technology as its predecessor the Vista X, but in a more compact form. Vista V features a meter bridge, motorised faders, and built-in Dynamic Automation with DAW remote control. The console is surround-sound capable, with versatile panning and monitoring functionality.

At the heart of Vista V is the Infinity Core, which uses CPU-based processors to deliver 800+ audio channels and more than 5,000 inputs and outputs.

The Vista V with Quad Star technology uses four processors to achieve 'aviation-standard' levels of redundancy in the control surface, while CPU-based DSP makes it viable to provide two completely independent DSP cores running in parallel with instant changeover, without a single sample of audio dropout.

Studer has announced the first MADI I-O card specifically designed to fit in the new D23 frame, which works with Studer Vista X and Vista V digital consoles. The card may also be used in the Vista 1 I-O card slot.

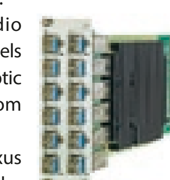
The new I-O card is a dual slot card fitted with two redundant MADI interfaces and provides up to 128 channels of inputs and 128 channels of outputs on the two fully redundant interfaces. Digigram's LX-IP AES67 PCIe sound card makes it possible for Studer's Infinity Core to integrate with audio-over-IP networks according to the AES67.

www.studer.ch

Stagetec

The On Air Flex console has a three-part structure in which its audio processing, user interface elements and control intelligence are separate components. By separating the component the mixer logic becomes freely configurable and available to all. The mixing console configuration can be defined with MapCfg software.

Like On Air 24, On Air Flex is a modular mixing console system with processing power provided by a Nexus XCMC board. It can process 40 input channels when configured with 8 groups, 8 sums, 8 auxes and 8 mix-minus buses. A further configuration with up to 54 input channels without groups is also possible. The input and output connections are available through the Nexus network, which can range from a single, locally installed Base Device to a campus-wide audio network.

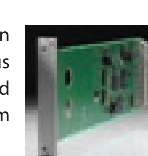


The new XRT board for Nexus audio networks provides more than 8000 channels of audio, and is equipped with 12 fibre-optic ports to route the many I-Os to and from the device.

In its latest upgraded incarnation, Nexus Logic Control provides Nexus users with the tools to realise far more elaborate controls in their systems than was previously possible. Users can define links between states and actions of the Nexus system and by using a variety of logic functions it is possible to query and link a broad range of states; this includes relay contacts and crosspoint states.

If a Stagetec mixing console is integrated in the Nexus network then its control signals also become available in the logic programming — for example, the information that a fader has been opened. It is also possible to trigger many mixing console functions, such as recall a snapshot, open an audio gate or control scene automation, and much more.

In the simplest case Logic Control turns on the studio red light when a fader is opened. In more sophisticated applications it enables rerouting of entire studios and production facilities, including all signals, at the push of a button. Benefits for Nexus Logic Control users include fewer external peripherals, lower additional costs and a reduced complexity of the overall installation.



Stagetec has introduced a new generation of XDSP signal-processing board for Nexus which quadruples processing power and integrates the Isostem upmix system from Dspecialists.

The XDSP board opens up new possibilities for audio signal processing within Nexus: up to 20 minutes delay, up to 66 30-band equalisers or up to 320 dynamic units. As before, all signal-processing modules can be combined individually according to the customer requirements meeting the demands of the most complex applications.

www.stagetec.com

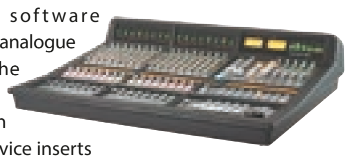
SSL

SSL's C100 HD Plus and C10 HD Plus digital broadcast consoles bring together new features and added value consolidation of SSL Production Assistant software to offer users 'the most powerful and flexible systems SSL has ever produced'.

The C100 HD Plus large format console is designed to offer a complete production solution for News and Sports in a standard high power configuration. Redundant Blackrock Processor cards in a 2U rack process 588 audio mix paths, with 256 channels of 6-Band EQ, and 284 channels of Dynamics with 512 channels of integrated MADI I-O.

The centre section penthouse has a 10-inch screen, which can be used to display any HDMI video source and enables the user to select from a wide range of metering options. The surface has been redesigned to facilitate clearer channel identification in a wider range of lighting conditions.

The C10 HD Plus has sizes from 16 to 48 faders and the self-contained fanless console is said to be ideal for all-round production demands in mid-scale broadcast facilities. The simple, button-driven surface is easy for new operators while the Blackrock Processor card offers 216 audio paths, with 160 channels of 6-Band EQ, and 188 channels of Dynamics with 512 channels of integrated MADI I-O. It also features a redesigned fader panel colour scheme.



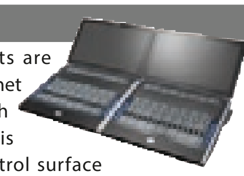
The integrated software controlled patching of analogue channel inserts in the original Matrix console has been upgraded in Matrix2. Hardware device inserts can now be loaded directly from the console hardware controls, previously this was done only via the remote browser software, with a new interface that facilitates loading individual processors, A/B comparison of different processors and building processor chains. The Matrix remote browser software has also been redesigned to provide a new 'drag and drop' style interface for loading processors and building chains.

A Fader Linking system has been added to the console, which allows two or more faders to be grouped, to facilitate stereo or 5.1 channel control or subgroup style mixing. The A-FADA (Analogue Fader Accesses DAW Automation) summing system used in Duality, AWS and the new SSL Sigma rack has been introduced to enable the analogue faders of Matrix2 to be driven by automation data from a user's DAW. A-FADA enables channel automation to be performed entirely in the analogue signal path but with the advantages of DAW automation data editing.

A collection of smaller new features have also been added including 'partial TR setup save and import,' which allows selected parts of the console setup to be saved and imported as setup templates; new Preset insert matrix 'scenes;' Preset insert naming tools; automatic dB readout for Pro Tools users, allowing the scribble strip to automatically display fader values upon touch; modifier key press and hold functionality for Cubase/Nuendo users and new DAW templates for Presonus Studio One and Ableton Live!

www.solidstatelogic.com

Salzbrenner



Polaris Evolution components are connected via a standard Ethernet network and recombined with each other for projects. Polaris Evolution consists of the control surface Polaris Access, the multi-user capable audio processor Polaris Scala, and the touchscreen upgrade Polaris View. Almost any number of these modules can be combined within an IP network, irrespective of physical location, and can also be used in parallel simultaneously for different mixing projects. This enables the customer to select the appropriate audio processing power for each application with the required number of fader strips and controls.

A single Polaris Access has 16 faders and the same number of dual rotary encoders, 48 buttons and a display screen strip across the width of the console. The Polaris View touchscreen upgrade can be docked at an angle onto Polaris Evolution and provides a convenient user interface. Polaris Access provides remote control of the Polaris Scala audio processor in the new mixer concept. Polaris Scala is a 19-inch unit for 256 audio inputs and 256 buses. Units can be cascaded to achieve larger numbers of audio channels when required.

Polaris Evolution is very scalable. If another Polaris module is plugged into the network, it is registered automatically. The user decides which mixing process it will be used for and whether to integrate it in parallel mode or as a supplementary device, all without any major changes to the configuration.

Just as it doesn't matter how many devices are logged into a computer network, it doesn't matter how many Polaris Evolution modules are deployed. You can connect them at any point in the network and use them individually or collectively as desired.

www.salzbrenner.de

Calrec

Calrec's Summa console for is live broadcasters who may not require as many resources as the company's Apollo and Artemis consoles. Its mechanical design keeps components and materials to a minimum and is around 30% more efficient than a comparable Artemis Light. Service access is from the front of the console and the entire control surface can be replaced by removing 12 screws.

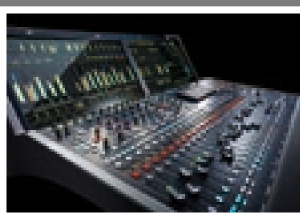
Users control the console via a 17-inch multitouch screen and the console simplifies such things as the creation of mix-minus feeds. It's other large displays are configurable to display bus, output, and loudness meters, and feature dedicated metering, routing, and processing information per fader.

The physical control surface is available in fixed 32- and 44-fader configurations. Each channel strip has a fader, two flexible control cells, and a dedicated gain pot.

It uses Calrec's Bluefin2 technology at its core and the same integral 8192 x 8192 Hydra2 router as the Apollo and Artemis consoles. This provides the console with a pool of 180 channel processing paths, which can be assigned as mono, stereo, or 5.1 channels. As with all Calrec consoles, there is no resource-sharing across the DSP, so all facilities are available on all channels at all times. It has eight groups, four mains (all of which can be mono, stereo, or 5.1), 16 auxes, and 32 tracks. The feature set also includes system redundancy, dedicated delay on all paths with additional assignable input and output delay, mechanical PFL overpress on all faders, and three 5.1 studio monitor outputs.

www.calrec.com

Lawo



The mc^{2}36 console is an all-in-one mixing desk with a feature set that makes it suitable to a broad range of applications. Its compact size belies its power as it has a DSP microcore with internal 512 x 512 port audio matrix and integrated I-O. As it is natively equipped with Ravenna/AES67, the mc^{2}36 integrates into IP infrastructures and for operational security the console has redundant power-supplies and DSP redundancy.

Its 21.5-inch HD touchscreens work with touch-sensitive colour-illuminated rotary encoders e.g. the dynamics window will automatically pop-up when touching the dynamics encoders, and after adjusting the parameters the auto-close function will close the window without additional user action to restore the full overview.

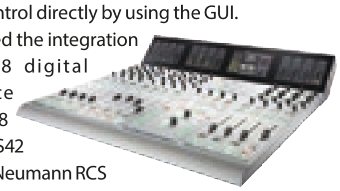
The mc^2 Compact I-O is a cost-efficient way to expand the mc^2 36's connectivity and to provide a distant stagebox solution in addition to the On-board I-O. Connected via Cat5 or fibre (optional), the ruggedised 5U stagebox provides 32 Mic/Line inputs, 32 Line outputs, 8 digital AES3 inputs, 8 digital AES3 outputs, 8 GPIO and a MADI (SFP) port. The mc^2 36 allows connection of three mc^2 Compact I-Os.

Software version 5.0 for all MkII mc^2 mixing consoles and routers represents an upgrade in functionality and operation.

Version 5.0 introduces the AutoMix feature, which automatically balances mono, stereo and surround sources for a variety of different applications including talk shows and panel discussions. The mxGUI is touchscreen-optimised software for monitoring and control of mc^2 consoles. Levels can now be adjusted via faders within mxGUI, making it a fully-fledged operating interface for the mc^2 series consoles and allowing improved remote control workflows and use as a backup.

Lawo's CrystalClear virtual radio mixing console optimises radio workflows by a smart console interface that is always aware of context, adapting to the skills of the user and the type of sources being used. Its control surface is software driven by a multitouch interface on a high-resolution computer display. Without physical knobs, buttons and faders, the virtual console presents the user with only relevant controls and information.

Software release 4.2 enables loudness metering for Crystal and Sapphire radio consoles. The consoles deliver loudness metering data for their main meters that may be based on mono, stereo, 5.1 and 5.1+2 signals. In addition, the Sapphire can display loudness metering for all input channels. In this case, the loudness metering is working in parallel to the channels' regular PPMs using metering elements in Lawo's VisTool touchscreen software. Users can select Momentary, Short Term or Integrated mode for measurement. In addition, the software update includes a graphical fader element, which has been implemented in VisTool to enable fader value control directly by using the GUI.



Lawo has demonstrated the integration of Neumann's DMI-8 digital microphone interface with Ravenna AOIP. DMI-8 supports eight digital AES42 mics and is operated via Neumann RCS remote control software that is integrated into mc^2 series consoles for adjustment of Gain, Pre Attenuation, Polar Patterns, Low Cut filter settings and others. The DMI-8 converts the AES42 format into Ravenna/AES67. Several DMI-8s can be cascaded and each digital mic interface can be addressed individually.

The mc^{2}56XT console offers the audio performance and features of the mc^{2}56 but doubles the fader count on the same footprint. The XT model can be configured with 48 to 144 faders arranged in a high-density dual fader layout, with further expansion achieved using 16 or 32 fader standalone extenders.

www.lawo.de

Radial

Radial's Space Heater is a combination 8-channel tube drive and summing mixer designed to bring character to digital recording.

It is an 8-channel 12 AX7 tube line amplifier with big fat Eclipse transformers that combine to produce a huge bottom end. A 3-position switch lets you select the applied voltage on the tube so that it can be low, mid or high fidelity.

Set up as four stereo pairs, the Space Heater design begins with a choice of ¼-inch TRS or D-sub inputs for quick connection to a workstation. Each channel pair is 100% discrete enabling four stereo sets to be used independently or sent to a stereo mix bus with left and right outputs. To control the effect on the tube, each channel set is equipped with a separate drive control to increase or decrease the signal being sent to the 12AX7 tube and a level control to set the output. A Heat switch lets the user apply 25, 50 or 100V on the tube depending on the fidelity needed.

www.radialeng.com

Soundcraft

Soundcraft's Vi3000 'all-in-one' digital live sound console has internal DSP Soundcraft SpiderCore, a new industrial design, 96 channels to mix, and onboard Dante compatibility.

DSP SpiderCore with Soundcraft's Vi Version 4.8 operating software offers the 3D Vistonics user interface while adding a fourth 24-channel fader layer to improve access to the console's 96 input channels. The surface operation and layout is similar to other Vi Series consoles but the Vi3000 features upgraded microphone preamps and 40-bit floating point DSP. It has 36 faders, 24 mono/stereo buses and a sweeping black screen panel with four Vistonics II touchscreen interfaces which allow it to be used by two engineers at the same time.

Roland

Roland's next-generation, live production digital console is the M-5000. Based on a new operating platform known as OHRCA, which stands for Open, High Resolution, and Configurable Architecture, the combination delivers 128 freely definable audio paths, a flexible user interface, expandable protocols and multiple format I-O choices. All paths are delivered at 24-bit/96kHz.

The key to the internal architecture is that it is not fixed to any configuration and can be freely defined within a range of 128 input or output channels or buses, allowing the operator to 'build' a console to suit the application. Each path can be used as a mixing channel, aux, matrix, subgroup bus or Mix-minus bus.

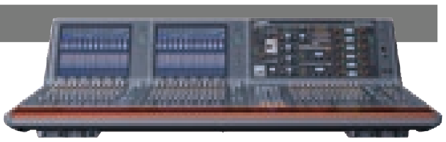
In addition to two REAC ports, the M-5000 has two expansion card slots to support Dante, MADI, Waves SoundGrid and other formats in the future. The back panel includes 16 x 16 analogue I-Os; 4 x 4 AES-EBUs; a 16 x 16 USB audio interface; connection for control via an iPad, connected or wireless; and control ports including footswitches, GPIOs, RS-232C and MIDI enabling the console to see up to 300 inputs and 296 outputs, all at 96kHz. Inputs can also be patched to outputs independently of mixing channel resources.

The control surface has a 12-inch colour touchscreen, 28 channel faders in four groups, multifunction knobs and buttons, 'selected knob' functions and a user assignable section. Displays are bright, colour-coded and visible in any light. The built-in GUI is expandable via Mac and Windows remote control software to any portable device.

The family of existing Roland products integrate with the new platform such as the digital snake options, the M-48 personal mixer and R-1000 48-channel player/recorder.

www.proAV.roland.com/OHRCA

Yamaha



Yamaha's Rivage PM10 is the company's new flagship large-format console and has newly-developed RY16-ML-Silk hybrid mic preamps partnered with 96kHz, 24-bit A-D convertors and Yamaha VCM digital modelling of Rupert Neve Designs transformer circuitry and acclaimed Silk processing. An engineer can have a completely transparent audio input path or, using the Silk Red and Blue modes and the Texture control in the console's selected channel, can be creative with the colour and character of each individual input.

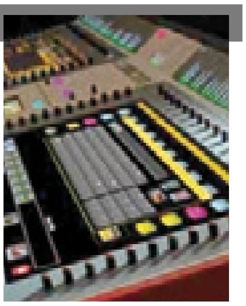
Twenty four of the control surface's channel strips extend virtually into two 15-inch touchscreen displays, while rotary encoders feature 'horseshoe' ring indicators. A third display screen can be added via a DVI socket, if required.

The backbone of the Rivage PM10 is Yamaha's newly developed Twinlane ring network, which can handle up to 400 audio channels at 96kHz, 32-bit over distances of up to 300 metres. Twinlane can connect up to eight RPio622 I-O units and, at launch, two CS-R10 control surfaces and two DSP-R10 DSP engines.

Yamaha's QL series of consoles comprises the QL1 and QL5. The QL1 features 16 inputs and eight outputs in a 468mm wide chassis with the QL5 32 ins/16 outs, measuring 828mm wide. The QL1 has 32 mono and eight stereo input channels, with the QL5 64 mono and eight stereo.

Designed for small-to-medium sized productions onboard Dante networking allows them to be integrated into bigger systems with Yamaha's R-series I-O units and CL Series consoles. Up to eight R-series units can be simultaneously controlled by a QL console, offering 256 input sources.

Cadac



V2.1 software for Cadac's CDC Eight console enables VCA group deployment, pressing of the Select button on a VCA deploys the VCA group members on the console surface. Switching between input-driven and mix-driven Fader Follow is enabled at a touch of the screen. It is now possible to view and access single input channel contributions to multiple buses, or a single bus's contributions from multiple inputs when in Fader-Follow mode. Features incorporated in V2.0 include individual channel section isolation and global automation safe in the console snapshot automation, as well as snapshot-specific recall filtering.

The CDC Four:m console has a 6 x 4 matrix and the M16 16-channel mic pre/MADI box. The matrix includes 31-band graphic EQ, compressor/limiter and delay on all four channels, and 31-band graphic EQ on all Aux sends.

Based around the user interface and audio performance of the CDC Eight, the CDC Six is less menu dependent than other consoles. The traditional fixed physical controls have been replaced by a user interface accessed via 23.5-inch touchscreen with encoders to the bottom and the right and 'touch and swipe' operation in which the faders follow the swipe of the screen or scrolling of the channels. CDC Six features stereo metering located to the left of each 100mm motorised fader, plus a full-colour OLED above each fader. The additional 6.5-inch LCD touchscreen located to the right of the main screen, provides rapid access to the advanced system controls and automation functions.

A 64-channel, 48 assignable bus design (as Group, Stereo Group, Aux, Stereo Aux or Matrix plus LCR, Monitors and Talkback), with 20 touchsensitive motorised faders, the CDC Six features Cadac mic preamps. It includes 4-band fully parametric EQ, 16 stereo on-board effects, extensive dynamics, input and output delays, snapshot automation, 16 VCA groups, with VCA unfold navigation, as well as compressor/limiter, 4-band fully parametric and 31-band graphic EQ on all outputs. It comes with an integrated Waves card for SoundGrid.

RCF

In a break from its traditional loudspeaker roots, RCF has unveiled the L-PAD range of audio mixers. The establishment of the RCF Mixer Division is said to represent an important strategic move and considerable investment in R&D.

The first nine-strong series of compact mixers provide flexibility for most live applications. The consoles in the RCF L-PAD series come with and without effects and are represented by 6, 8, 10, 12, 16 and 24-channel versions.

Models have compressors with a single knob that controls the threshold and compression ratio at the same time, internal multi-effects, 'warm–sounding' and transparent mic preamps, 3-band EQ on mono channels and the 2-band EQ on the stereo channels. The L-PAD8C, 8CX and 10C have 45mm faders while on the 12C and 12CX models have 60mm faders. DSP takes care of the management of the internal effects — 16 presets are available in the RCF L-PAD6X while L-PAD8CX and 12CX have a choice of 99 different presets.

Designed as a practical fader-free portable mixer the M18 digital WiFi mixer features an on-board WiFi access-point with integrated internal antenna, enabling all functions to be controlled wirelessly.

The tablet mixer sports a suite of plug-in algorithms and effects, including several equaliser types and classic emulations of guitar and bass amplifiers, as developed by plug-in developer Overloud. Targeted at musicians and small combos rather than sound engineers it has two high-impedance lines and a range of effects.

The 4-band EQs on each channel feature three different modes: Standard, Vintage and Smooth, allowing a wide range of EQ sounds and tweaking capabilities. Up to 15 simultaneous insert effects can be inserted on the inputs without the need for additional equipment.

The M18's back panel has eight Mic inputs (with six XLRs and two combi connectors), 10 Line inputs, six Aux Out, Headphones Out and two XLR for Main Out. In addition to the six Aux buses for stage monitoring or any other external effects, three separate aux buses are available to feed the internal effects.

www.rcf.it

Spotlight: Fairlight

Fairlight has reinforced its drive towards cinematic immersive sound delivery by launching a platform for Object Oriented 3D Sound Production.

This new platform supports traditional standard and custom busing in simultaneous 2D and 3D, along with NHK's 22.2, Dolby Atmos (via RMU) and DTS MDA. It also supports third party client applications such as Avid Pro Tools 10.x and Nuendo 6.0.

The fifth major software release for the second generation CC-2 FPGA audio engine is said to make it perfectly suited to the demanding delivery requirements of today's broadcast industry. By providing more than 1,000 playback channels, 100 live inputs and 100+ output buses, the system is claimed to be the most powerful audio engine made.

Fairlight has combined its open platform Media GateWay with its iCan workflow automation tools. These Gateways offer 'a glimpse into the future of broadcast postproduction' by embracing collaborative editing environments such as Quantel's Qtube, with internet-based video source reference, original source audio and faster-than-real-time layback.

Fairlight has implemented native Auro-3D mixing with the Auro-Codec integrated in Fairlight's 3D Audio Workspace (3DAW). 3DAW adds immersive sound capability to existing DAWs while remaining compatible with established workflows.

Fairlight's SX-36 is designed as a high-end I-O for audio production systems powered by Fairlight's Crystal Core Media Engine. It offers a variety of analogue and digital I-O, remote controlled mic preamps, precise lock to timecode and low latency.

It has two remote-controlled mic/instrument preamps, 8 balanced analogue inputs, 12 balanced analogue outputs, 9 stereo digital inputs and 11 stereo digital outputs, one stereo digital input with sample-rate conversion and one stereo digital output (SPDIFF).

Fairlight's Xstream and latest generation Xynergi controllers have extended their control capability to non-linear video editors such as Adobe Premiere Pro, Grass Valley Edius and Sony Xpri NS. The control surfaces consist largely of Fairlight's unique picture keys, which self-label instantly for each task performed, displaying the right commands and functions at the right time.

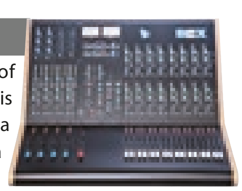
New iCan (Integrated Control Across Network) technology provides the operator with a layout editor to design customised keyboard layouts for the video editing software in use. The system can record and play back macros and multiple applications can be accessed simultaneously. Also included is an Xplain help function and support for 18 different languages.

Extending its range of dual purpose, production consoles, Quantum.Live can switch between live and postproduction at the touch of a button.

It is available in a Table-Top configuration that complements the Evo.Live family and comes with 12 faders accommodating 144 signal paths over 12 layers. A second TT frame can be added, increasing the system to 24 faders. Quantum.Live can be customised with tools such as Fairlight's Smart. Cart sound FX player, the ability to integrate soundbeds playback, AFV support and offline session prep.

www.fairlight.com.au

API



API's latest addition to its line of analogue consoles — The Box — is designed for those who require a smaller format console but with a big console sound. The Box features the same circuitry, performance and API sound as the company's Vision, Legacy Plus and 1608 consoles.

'The Box offers an easy, turnkey solution for recording and mixing,' said API president Larry Droppa. 'It's a great option for people who record a few channels at a time, but demand the warmth and punch that a large API console delivers. In addition to four inputs, full centre section control, and 16 channels of API's famous summing, the icing on the cake is a classic API stereo compressor on the programme bus. Now you can truly record and mix — in The Box.'

www.apiaudio.com

Harrison

Harrison Consoles' newest analogue desk is the 950mx. Intended for facilities that need analogue monitoring, mixing, and summing when working with a DAW, it provides a large-format console sound and construction while forgoing the multitrack buses and inline monitoring features that are less necessary with DAW workflows.

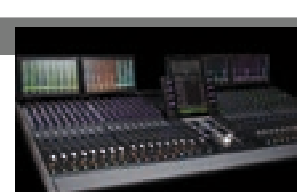
The 950mx features a robust ground plane, gold module connectors, gold-plated switches, conductive plastic knobs, and fully-differential balanced I-O at every point. The summing buses are carried via PCBs, not ribbon cables while a custom-designed linear power supply provides 'rock-steady voltage for clean sound, robust EQs, and generous headroom'.

All 950mx mono input modules now feature individually switchable, sweepable high-and-low pass filters and 3-band sweepable EQ — featuring the same circuitry as the original Harrison 32-series consoles. Mono input channels also have switchable inserts, a mic preamp (switchable to a line input), four aux sends, 100mm P+G faders, two Mix Bus assignments, and a postfader direct output. Stereo input modules have switchable high and low pass filters, 3-band tone controls, balance, channel reverse, mono sum, input trim, four aux sends, 100mm P+G faders, and two Mix Bus assignments. Another user-requested feature was the addition of an alternate speaker output.

Mono and stereo modules of the 950m and 950mx are interchangeable and 950m users can arrange to have their mono channels updated to the reflect the new features of the 950mx.

http://analog.harrisonconsoles.com

Avid



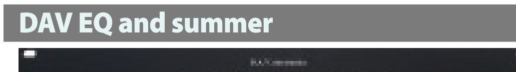
The Pro Tools|S6 Master Joystick Module features dual non-motorised, touch-sensitive joysticks for surround sound mixing with Pro Tools|S6. The module is an option for M10- and M40-based S6 systems, and includes a 3.2-inch TFT display that displays current pan and joystick locations. The module also includes Strip Control and Knob Cell sections for each Joystick which are similar to those used on the Pro Tools|S6 Knob- and Process-Modules. These feature two multicolour LED top-lit encoders, four OLED displays for visual feedback and panning control switches for controlling LFE, divergence, and more. Plus, 16 panning mode switches with colour LEDs for enhanced panning.

Venue|S6L is a modular, scalable live sound mixing system for FOH, monitor, broadcast, and theatre. It offers Pro Tools recording and playback capabilities, including Virtual Soundcheck, without needing an external interface and can share I-O across multiple networked systems with Avid True Gain advanced gain tracking technology. Visual feedback and access is through high-resolution, daylight-visible touchscreens and OLED displays. It can interface with a range of network and I-O formats, including Ethernet AVB, Dante, MADI, and Thunderbolt.

Avid System 5 version 6.0 software provides support for and integration with Dolby Atmos and includes direct Atmos plug-in control in Pro Tools from System 5. There's also deeper integration with Eucon control including VCA Spill, and automation control. Pro Tools | S6 software version 1.2 offers VCA Spill (ability to spill VCA slaves onto the surface from a VCA master to access and update as necessary), Multi-Workstation Layouts, Expand Mode, Audio Editing from an S6 Channelstrip and Gain Reduction Enhancements. There's also Integrated DVI Switcher Control, Automation LED indicators on knob modules, and a Soft Key Editor.

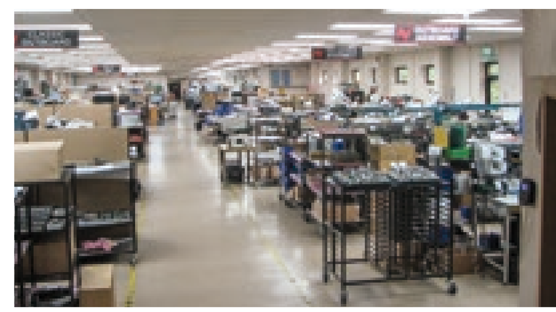
www.avid.com

DAV EQ and summer



The DAV Summing Mixer is a passive 32-input summing mixer with inputs on 4 x DB25 connectors and stereo outputs on XLR.

www.kmraudio.com



AMS-Neve

Mark Crabtree, the man behind the AMS-Neve brand, talks technology and the Made in Burnley ethic. ZENON SCHOEPE

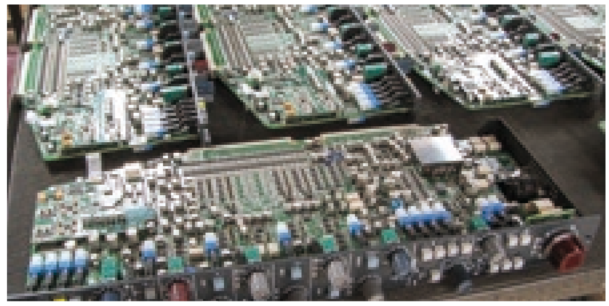
AMS-Neve has had an 'interesting' history yet it's one that has always been tied umbilically to the fate and development of our industry; and that's because it has been pivotal to progress. Without putting too fine a point on it much of what we have taken for granted forever — how our equipment looks, reacts, behaves and sounds — was defined long ago by the separate and combined contribution of the constituent brands in the title. For many the Neve brand remains the pinnacle of analogue excellence and desirability. For others, AMS trail-blazed at the true bleeding edge of the steam-driven, formative years of digital and its legacy is seen everywhere today.

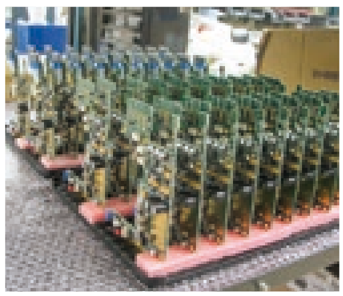
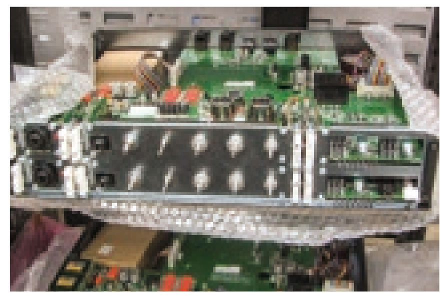
But the history was 'interesting' and not without considerable challenges. There were various corporate incidents for AMS along the way including a merger with Calrec, which ultimately led to a divorce but managed to leave behind some important technological milestones. There was the purchase of AMS and Neve by Siemens which then forced the combination of the two brands into AMS-Neve — and their completely different cultures, ideologies, technologies and markets — before Siemens extricated itself from the arrangement and walked away. What followed were years of shifting ownership and outside intervention and some difficult times. Despite this, at all stages there has been an unmistakable and continuous contribution to the pantheon of audio technology. At all stages there has also been the consistent contribution and involvement of Mark Crabtree.

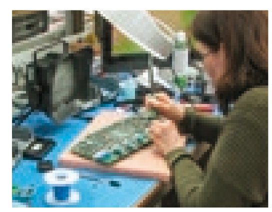
The co-founder of AMS was tied to the mast of AMS-Neve with the merger and since 2011 has been the sole owner of the company in what has been a revitalisation process. Mark is clearly enjoying himself at the helm of a modern, focussed and innovative company. He's even had time enough to put some of his boundless energy into the promotion of his home town of Burnley, including encouraging an interest in practical science in local schools through a scheme with other local manufacturing businesses. Among other things, schools have been making their own electric cars in a move to inspire budding engineers and inventors. They're Making it in Burnley — just like all AMS-Neve products — and Mark is adamant that 'You can make a difference.' It's work that has seen him awarded an OBE.

Mark is keen to put the product range into context. Those with long enough memories will remember that AMS-Neve digital products always enjoyed an evolutionary process that added power and functionality as user demands changed. The DFC Gemini film console is a case in point. 'It's continually changing internally and externally,' says Mark. 'We've done some work with Dolby for Atmos and we natively support all the 3D formats. And users are finding they need more and more power — they're using 2000 paths in a mix now which is quite staggering — and we have the most powerful processing anywhere in that market.'

The analogue 88RS and the SP2 scoring panel for 127 buses are still attractive products for a specific applications but the broadest analogue desk appeal is the Genesys in standard and Black. 'Genesys was always meant to wrap around any workstation and it was always intended that you would be able to plug into a Mac and then you'd got yourself a studio,' he says. 'We spent a lot of time getting all the classic 1084 EQs being digitally controllable — we also have the 88R — and these are effectively hardware plug-ins for the desk. For the basic desk, if you want to work with a workstation you're going to need some faders and mic preamps, some meters, some monitoring, all those things you need and can make out of a bare-bones Genesys. From then, you can automate the Genesys, have complete reset, the ability to plug-in different sorts of dynamics and EQ, you have the option to put MADI cards in the back, and so on. It wraps around any workstation and you've got the quality — and possibly more features — of the 88R in a compact and affordable frame. It's meant to be as seamless as possible — its codename was 88KI for Killer Integration.

'That's gone very well but I had always envisioned even more integration with the workstation,' says Mark reminding that it was AMS that first came up with the idea of an editor combined and integrated into a console. 'Now there are so many different flavours of workstation and everything is going touch so we put a touchscreen in the middle of the Genesys and linked that very cleverly to its in-built computer, to the network, to the workstation, to remote control of mic pres — if you want to put the 4081s on there they hook-in seamlessly. The whole thing is a joy to plug-in and get going with and that's the Genesys Black. The Black also covers postproduction and in its entry-level guise it's got 16 inputs, 8 channel strips with inline, the returns, so you have quite a lot of inputs there but you can expand it up to 64 strips.

'It takes people a while to catch on what we're on about — people say it's a bit like this, like that and the other. Well it isn't; it's a bit like itself. It has a personality now and traction all the way from post to a main studio console where you can have a 64-fader, 128-input Neve. It doesn't take the power of the big desks and it integrates with workstations. It's also fantastic for schools.'

The outboard range — referred to as classics and new classics — has introduced a new generation to the Neve brand. The classics are hand wired exactly as they were in the 1970s

'Ours are the only real ones. Only we know the full spec of the transformers,' explains Mark. 'We've gone back to the roots of the transformer and we've connected up with the people who used to make them a long time ago and these are full Marinair–spec transformers and they are the real deal.

'We have the hand-written original drawings — John Turner has been with Neve for about 50 years and Robin Porter has been with Neve for over 40 years — there's a wealth of original information, all the hand written specs. We've made sure that when it says 1073 on it, it is a 1073 — no ifs, ands or buts.' The desire to broaden appeal of what are quite exclusive products came via the 500 series format.

'We took quite a bold step at the time with the 1073 LB,' he says. 'It's been a massive hit and has sort of allowed users' ears to be educated on a much wider basis. There are those who plugged one into their Lunchboxes and couldn't believe they could sound that good. How can you know unless you've heard it?'

As a dyed-in-the-wool early-adopter technologist Mark has a very individual appreciation and attitude towards digital processing. AMS-Neve uses Sharcs for its DSP but he is fully aware of alternative approaches such as the CPU route. 'We've been through lots of different types of DSP, we don't really care what it is so much as how it sounds and how it's programmed,' he says. 'There are some restrictions with CPUs in that if you were trying to get 2000 inputs in and out of a CPU you've got problems — they will ease off with time — but bottlenecking in and out of the motherboard is a problem. The fact that motherboards keep changing is also a problem; people buy our stuff and expect it to last 15 years without doubt and it does.'

From experience he says that relying on third parties for crucial parts of your product means you will be in for a long ride. 'PC motherboards are the same; if suddenly drivers don't work anymore. There is also something weird afoot because clearly the way PCs are going the number of people buying full-size motherboards is getting fewer and fewer. Think about it, everyone is buying tablets so the need for doing a standard PC motherboard that is open for everyone is going to drop and that means that the prices will go up and then you'll be down to a few developers. We've seen that with electromechanical components — the number of people who use proper switches these days with gold on them is fewer so it's become a specialist niche business. I think that's the way the motherboard business is going to go too.

'What will be interesting with workstations — as the world is moving towards touch — is that the whole of the interface of workstations will have to change radically. If you see what's happening with Microsoft and Apple they are trying to homogenise across tablets and PCs. So that nice cosy world of mice and PCs is slowly becoming more of a specialist field. It will be very interesting to see what happens with the mainstream workstation people who really take on touch in a big way.'

Mark is unusual as a company head in that he designs gear, understands it and how it is used, understands what it takes to run (and to rescue) a company, is intimately involved with the manufacturing process and cares about its optimisation, productivity and the well-being of the team behind it all. AMS-Neve employs about 64 and aside from some bare fibreglass and metalwork that is bought in everything else is Made in Burnley.

The factory is arranged in sections with the machines — stuffing, soldering and testing — at the back of the plant leading into the various stages of sub assembly, testing and final assembly.

Mark's involvement with the manufacturing is demonstrated by how he helped optimise the component stuffing machine from the stop/start nature of producing a large selection of products that require comparatively few and different board types.

'There is such a mix of things that have to go through that I set out to try and optimise the machine because you don't want to have to tear down all the component trolleys every time you want to make a different product. I worked out a methodology of how to put the boards in the right order so you have to change the fewest number of components. I knocked up a huge spreadsheet over a weekend of how to do it and my son did some database conversion to plug it into our manufacturing resource planning software.'

I suggest that there aren't that many company bosses who could do this 'or are crazy enough,' he interjects, 'or interested enough. That was an interesting challenge and again it all goes around in a circle because we could spend another half a million pounds on another machine but I could see that there was a better way of running the one we have.'

He says he runs manufacturing 'like a flock of birds'. 'Rather than having people asking "haven't you finished that yet, are you doing this?" We've got a system now where they get a sheet of paper at the beginning of each week telling them what we'd like them to do in any order they want. And they do and just get on with it. They move around, they multiskill into what's required. We have a nice flexible workforce and there is satisfaction in making things — making a 1000 switches an hour is hard — but making this stuff is interesting and important.'

Contact

AMS-NEVE, UK
Website: www.ams-neve.com