

THE SPIRIT GUIDE TO DIGITAL MIXING

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THE BASICS OF DIGITAL MIXING

Welcome to the Spirit Digital Mixing Handbook – the essential guide to the world of digital mixing. Over the years it has become increasingly easy for the solo musician, producer or small band to assemble a small studio, and with the advent of affordable digital technology, many of us now have access to our own sophisticated recording facilities capable of producing release-quality material. As the central, and perhaps the most complex, single piece of equipment in the studio, the mixer has always been the most important item to get to grips with. Hence this guide, which helps you understand the principles and practice of digital mixing. Enjoy!

A. Analogue & Digital – What’s The Difference?

The move from analogue to digital technology in the world of audio is a revolutionary one, not just a small step. The technology is fundamentally different, hence many of the the advantages of digital over analogue. Although most practical aspects of recording and mixing are unchanged, some are not, and it helps greatly to understand some of the principles of digital audio.

Whereas analogue electronics deal with continuously variable quantities (such as voltages and currents), digital electronics deal with discrete values - numbers that can take only one of a predefined set of possibilities. Digital circuits do this by dealing with 1s and 0s, represented by the difference between a switch being on or off, or a voltage being present or not.

This difference means that digital and analogue systems represent sound in very different ways. What we perceive as sound is in fact a vibration in the air. Hit a cymbal and it vibrates, transmitting the movement to the air around it, and as the vibrations are passed on to more and more air

molecules a series of pressure waves are created. As long as the frequency of these vibrations is within the range of human hearing (roughly 20 to 20,000 cycles per second, depending on your age), we hear them as sound waves, and when they reach our ear drum we hear the cymbal. But if we want to do anything more with this sound – record it, for example – we need to turn it into something that we can manipulate more easily than pressure waves in air.

In analogue audio circuits, sound waves are represented by continuous electrical signals. At one end of a typical audio chain, a microphone uses the vibrations of air against its diaphragm to generate a varying electrical current, and at the other a speaker uses a varying electrical current to produce vibrations in the air next to its cone.

Analogue electronics are always prone to noise – at the very least because the the random thermal motion of electrons in components will produce some noise – though this can be minimised through good circuit design, the use of high quality components, and good construction.

In digital equipment, the audio waveform is represented by a series of numbers. An analogue signal can be turned into digital form with an analogue-to-digital converter, which measures an analogue input voltage at regular intervals and generates a sample (a number that represents the instantaneous strength of the signal) each time. A converter that runs at a sampling rate of 44.1kHz samples the signal 44,100 times per second, and in a well-designed converter the size of the samples should be in direct proportion to the input voltage.

A 16-bit converter can resolve 65,536 levels.

A 20-bit converter can resolve 1,048,576 levels.

A 24-bit converter can resolve 16,777,216 levels.

Because numbers can be described and copied with absolute precision, digital audio signals can be passed from one machine to another, or from one part of a digital mixer to another, with no change to the sound. An analogue signal, on the other hand, cannot be passed through any component without

affecting it slightly. An analogy might be building a model out of clay or out of Lego™ bricks. You can make copies of both, but even with the best casting methods, you won't be able to make a perfect copy of the clay model. No matter how many times you make copies, or copies of copies of the Lego™ model, each can be exactly like the original.

One of the first areas of the audio to benefit from the introduction of digital technology was recording and playback equipment, and digital recorders have now almost entirely replaced analogue recorders in every application. Apart from the fact that a well-designed recorder with good converters offers very high audio quality, they offer two distinct advantages. Firstly, the quality of digital recordings remains unchanged even after repeated playback. Analogue tape recordings, on the other hand, shows a progressive drop in quality after repeated passes. Secondly, digital recorders can use random-access rather than linear recording media, which allows them to play back recorded material in many ways other than simply 'as recorded.' You can repeat sections (for choruses, for example), perform cut-and-paste editing operations, and so on.

328 FOCUS: CONVERTERS

Converting between analogue and digital is one of the most critical stages of an audio signal path. Once a signal is in digital form, it's fairly robust, but converting to and from analogue offers a chance for audio quality to degrade significantly. In order to capture input signals as faithfully as possible, analogue-to-digital converters should offer high bit resolution – 16 or 18-bit converters are looking rather outgunned in these days of 20 and even 24-bit converters. The Spirit Digital 328 has 24-bit 128-oversampling A-to-D and D-to-A converters, with considerably greater resolution than established media such as CD and DAT (16-bit), and allowing use with 24-bit systems such as newer PC-based recording platforms.

At the output stage, the 328's conversion is also 24-bit with 128-times oversampling.

B. Bits and Sample Rates

The two most important factors determining the quality of a digital audio system are sample rates and bit resolution. A properly designed digital recording system can reproduce audio frequencies up to half of its sampling rate. So, in order to cover the full range of human hearing, a sample rate of 40kHz is required – hence the two most common sample rates of 44.1kHz (the CD standard) and 48kHz.

Bit resolution, on the other hand, determines the signal-to-noise ratio of the system - that is, how much difference there is between its quietest and loudest output. You may have heard that digital systems can achieve a performance of 6dB of signal-to-noise per bit, so a 16-bit system (CD or DAT, for example) manages 96dB. A 20-bit system can in theory manage 120dB, though in higher resolution systems it becomes harder and harder to achieve theoretical performance.

328 FOCUS: USING DITHER

When you take a 24-bit audio signal down to 16-bit resolution, you inevitably lose quality. Whereas the 24-bit signal could have a signal-to-noise ratio of 144dB, the 16-bit version of it has a signal-to-noise ratio of 96dB. The Spirit Digital 328's stereo digital outputs, both AES/EBU and SPDIF, use a technique called dithering to preserve as much quality as possible when going down to lower-resolution media, such as DAT, which have poorer signal-to-noise performance than the mixer's internal signal path. Dithering involves adding a small amount of noise to the 24-bit signal at just about the theoretical noise floor of a 16-bit signal. Odd though it may sound, this trick means that you can hear low level sounds (reverb tails, for example) well down below the noise floor, and the 16-bit signal actually achieves a dynamic range of around 115dB. This improvement is audible no matter what DAT player the recording is played on - it does not require any special decoding.

C. The Question Of Levels

In the world of digital recording, you have to treat levels with a little more respect than in analogue systems in order to avoid distortion. This is because of the very different ways in which analogue and digital systems work.

We know distortion when we hear it, but how does it arise? Put simply, distortion is an unwanted change in the audio waveform, usually caused by the clipping of the audio waveform. As you crank up input levels, you reach a point where the system cannot recreate at the output the extreme peaks and troughs at the input – and that’s distortion. In analogue circuits distortion tends to come on gradually, increasing in severity as input levels are increased. Because it can sound good to the human ear, some types of analogue distortion have long been exploited in the studio – to warm up drum sounds by recording ‘hot’ to tape, by using tape saturation characteristics. So, although the point at which distortion begins can usually be quite well identified, you can generally ‘push’ this level a little without ruining your recording.

Digital electronics, on the other hand, deal with a precisely defined range of levels. As you increase the level going into an analogue-to-digital converter, the numbers coming out get bigger and bigger – until you run out of resolution at the converter’s maximum input level, and you have the sudden onset of distortion. For reasons which are too tedious to go into, digital distortion doesn’t sound at all pleasant, and it’s therefore very important to avoid exceeding maximum levels.

Metering also works differently. In analogue mixers and recorders, equipment is designed to operate at various operating levels, such as +4dBV (most ‘professional equipment’) or -10dBu, and the relationship between 0dB on meters and the point at which distortion starts is not fixed. Also, because distortion comes in gradually, you can almost always get away with pushing your meters some way into the red.

In a digital system, metering is relative to 0dBFS, which is the point at which the system has run out of numbers and you cannot go any higher without creating distortion. There is no LED on a bargraph meter above this, because there isn’t anywhere else to go. Because distortion occurs as soon as you exceed these levels, whenever you’re setting levels bear in mind that even peaks should not light the red LEDs at the top of the 328’s meters, though this might feel right if you’re used to working with analogue mixers.

D. The Future

Because digital audio technology rides benefits from progress in the very fast moving area of computer technology, things are always moving ahead. From 16 bits and 44.1kHz, we’ve moved on to a point where 20 and 24-bit recorders are in use, some with 96kHz sampling. The need to move greater amounts of data around means that new interface standards will appear. Firewire (IEEE1394) is already establishing itself as a useful transfer protocol in digital video, though it may soon also be used in audio. Another trend that is likely to continue is the use of cheap, powerful consumer or business computers to run increasingly powerful audio software, software that can take over more functions previously performed by dedicated hardware.

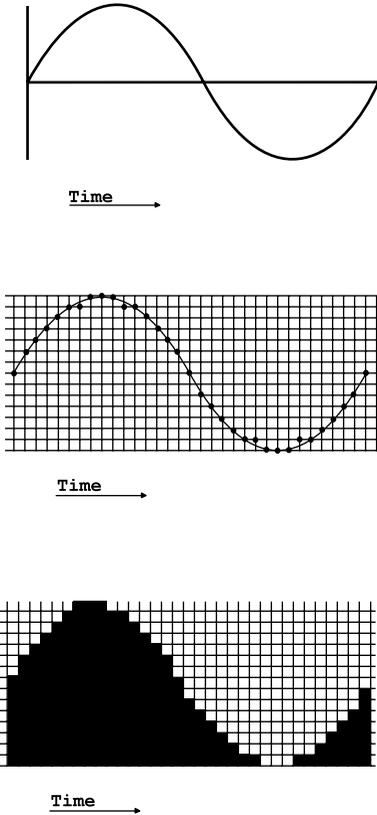


FIG 1.1
Waveforms as represented in analogue and digital systems

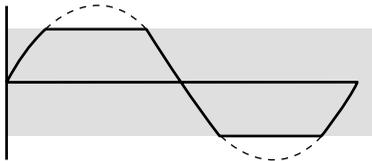


FIG 1.2
Distortion in analogue and digital systems.

328 FOCUS: AUDIO QUALITY

Spirit has earned a reputation over the years for very high audio quality at affordable prices, and the Digital 328 carries on that tradition. As well as its high-quality converters – see “Using Dither” on page 4 – the Digital 328 also offers very high resolution (56 bits) in its internal processing. This is important because everything that happens inside a digital mixer – combining two signals, changing a level, deriving a feed to an external Aux or internal FX send – involve making calculations, and these calculations must be carried out with as high a degree of precision as possible. Each time a calculation is performed, you need a longer wordlength to store the result without ‘rounding off’ the result, and thereby losing precision that carries valuable information about the sound. So, although the Digital 328 starts with 24-bit numbers at one end, and ends up with 24-bit numbers at the other, using 56-bit precision for the processing along the way means that repeated processing steps can be applied without adding noise or distortion.

THE BENEFITS OF DIGITAL MIXING

Although on the inside the way that a digital mixer works is quite different to the way that an analogue mixer works, they do the same job – combining your various audio sources, mixing and controlling them on their way to whatever outputs you need. In some applications, the advantages of a digital mixer are directly connected with the way that it works inside, in other applications the advantages are less obviously connected to its digital nature.

A. Keep Music Digital

In a typical audio chain, the sound of an instrument or voice may pass through a mixer several times before it makes it way to a pair of speakers or headphones – once on its way to a digital recorder, then back through the mixer again during mixing, out and back in via an effects loop, and so on. Now that so much of the equipment around the mixer is digital, if the mixer offers only an analogue signal path then the audio may go through many steps of conversion between analogue and digital, losing a little quality each time. With a digital mixer, however, you can transfer audio to and from your digital multitrack, and to and from digital effects units, entirely in the digital domain. Many synths and samplers now offer digital outputs, and a digital mixer can also add effects, and perform dynamic processing, all without going back out to analogue. By cutting out unnecessary conversion steps, your recordings can stay in the digital domain from the moment they come into the desk to the moment they come out of a CD player, ensuring the best possible quality.

B. Take Control Of Your Mix

One of the basic features of analogue mixer construction is that all of the hardware controls are also audio components. When you move a channel fader, you're changing the resistance in an

audio component in the signal path. When you move an EQ knob, you're tweaking a variable pot that is part of a channel strip's EQ circuitry. With a digital mixer things work completely differently. The audio processing takes place in chips that are not directly linked to any controls, which presents a great opportunity for designers to create new interfaces for their products, freed from the slavery of one-control-per-function. Because the audio handling and control aspects of the desks are now separate, each can be made to work as well as possible, rather than there being any compromise. And contrary to popular belief, there's no need for a digital mixer to behave like a computer, relying on menu-driven interfaces rather than intuitive controls.

Although it would be possible to create a digital mixer with a control for every parameter, such a design would be much more expensive than one in which fewer assignable controls are used. The Spirit Digital 328, for example, uses clearly laid-out assignable controls, and behaves like a traditional mixer rather than a computer. All major desk operations can be achieved with a few simple and intuitive control inputs, and you hardly ever need to refer to the LCD screen. Nonetheless, you have the flexibility that comes with controls that can change at the touch of a button – one minute you're adjusting levels, the next you're sending out MIDI controller messages. And if you want to go beyond the 328's own control surface, you can use any major sequencer program running on any computer platform to map the Spirit 328 controls to on-screen graphic controls.

Because a layer of software sits between the physical controls and the audio processing of a digital mixer, it is relatively easy to add advanced features such as automation and remote MIDI operation, all of which gives you more mixing control. The two types of automation that you find on digital mixers are 'snapshots' – which store all the desk's audio settings and can be recalled at the touch of a button or against timecode – and dynamic automation, in which you can record and playback any and all of the fader and other control changes that you make.

328 FOCUS: DIGITAL INTERFACES

The Spirit Digital 328 has both stereo and multi-channel digital interfaces, enabling you to connect a wide range of digital equipment to its inputs and outputs. There are two sets of 8-channel ins and outs, using on-board ADAT optical and TDIF (Tascam) format interfaces, for group/tape outputs and tape returns. Both AES/EBU and SPDIF interfaces are also supported, allowing FX sends and returns, the stereo mix output, and 2-track returns (for DAT machines or other stereo mastering media) to be carried by digital as well as analogue connections.

328 FOCUS: USE WITH PCI CARDS

More and more studios, especially small personal facilities, exploit the power of PCI cards installed in personal computers to provide cost-effective yet high quality and sophisticated recording and production. A fairly standard PC or Mac, equipped with a suitable £500 card, could offer multi-track random access digital recording and playback, replacing a studio multitrack tape recorder. With the right card and software, you could add sample playback and sophisticated effects, putting many of a studio's electronic instruments inside the computer. This technology has become very cheap because of the falling price and easy availability of chips and other components. Also, the adoption of multi-channel digital interfaces like the ADAT optical system means that the ability to handle multiple channels of audio input and output can be easily included on these cards, but a good mixer front end is essential to exploit this kind of technology to the full. The Digital 328 interfaces easily and effectively with these cards and the systems that they are intended to support. A typical setup might involve a Mac or PC with a Korg 1212 card, running Steinberg's Cubase Audio VST software. Using the ADAT optical connections, the 328's eight group outputs feed the Korg 1212's eight ADAT inputs, and the 'tape' returns go from computer to the 328 also via an ADAT optical interface. (If the computer is fast enough to allow

more than eight tracks of playback, the internal pre-mixing abilities of Cubase Audio can be used to pre-mix several tracks to each of the eight available output busses).

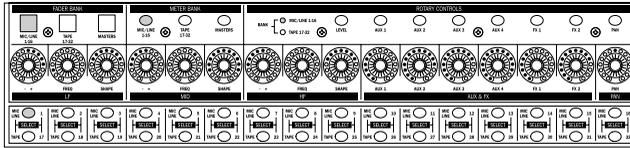
328 FOCUS - SNAPSHOTS VERSUS DYNAMIC AUTOMATION

Depending on the nature of a project, either the 328's snapshot or dynamic automation, or a mixture of both, may be most appropriate. For example, you might use snapshots to store basic mixer setups for several different songs that you are working in, or for sessions for different artists, and then use dynamic automation for mixdown. If you're working to picture, snapshots can be particularly useful. By recalling snapshots against timecode, changes in the mix can be easily made to happen in precise sync with picture edits, and if the picture is re-cut, you simply need to change the timecode values for the audio cues rather than rebuild a whole dynamic mix.

328 FOCUS: E-STRIP

The key to the Spirit Digital 328's user interface is the E-strip, a row of 16 rotary encoders above the 16 motorised channel faders. Its primary function is to operate exactly like a regular analogue mixer channel, only rotated through 90 degrees, with the familiar 3-band EQ controls, four aux sends, two effects sends (to the two internal processors), and pan control. As well as being familiar, it's easy to use: just hit the select button for the channel you want, and the E-strip's continuous rotary controllers put all the channel EQ, aux and pan facilities at your fingertips. A ring of LEDs around each controller shows you the current setting for each parameter. Channel features that are used less often, such as bus routing and phase reverse, are controlled via the master section, while the channel's analogue controls (100Hz high-pass filter and channel input gain) are found at the top of the front panel. The second function of the E-strip is to give simultaneous control of a set of 16 Level, Pan, or send controls for FX1 or 2, or Aux 1 to 4, for channels 1-16 or 17-32.

FIG 2.1
E-Strip



328 FOCUS: QUERY SYSTEM

Because the Spirit Digital 328 is a very flexible and fully-featured mixer, it's simply not possible to show all desk parameters simultaneously in order to let you see how everything is routed or switched – that would produce a very confusing front panel, whereas the 328 design approach has been to create an uncluttered, easy-to-navigate interface. On an analogue desk, you can quite easily check the desk state “at a glance”, because there’s one control per function – the trade-off is that you can end up with a huge number of controls on a front panel. The idea behind the query system on the Spirit Digital 328 is that it allows the user to ask a simple question and get a simple answer. The key is that it uses the same intuitive means to check routing as to make the routing assignments. To route channels 3 and 4 to the stereo mix bus, for example, you do the following: Press the Select button for channel 3, and see if the Route To Mix button is lit. If not, then press the Route To Mix button to assign the channel to the mix bus; the button lights up green to indicate that this has been done. Now press the select button for Channel 4, and repeat the procedure. If you now want to check which channels are routed to the Mix bus, press and hold the Route To Mix button; all channels which are routed to the mix will be shown by their illuminated Select buttons, while all channels that are not will remain dimmed. With this one button push, you can check all 32 mono channels and tape returns, both stereo inputs, eight groups and both FX returns – and to make routing assignments easier still, you can add more channels to those routed to the mix by hitting their dimmed Select buttons while you keep the Route To Mix button held down.

328 FOCUS: FADER LAWS

Another aspect of a digital mixer’s user interface that can be more easily modified or controlled than

on an analogue console is the fader laws – that is, exactly how your control inputs translate into an audible effect in the signal. For example, on the channel, master and group faders of most mixers – including the 328 – movements in the top half of the fader produce less audible change in level than the same movement lower down, which means that in the crucial area around 0dB gain, where you effectively do most of your mixing, you have most control.

On the 328, the response of all controls is carefully tailored to suit the parameter you are changing. When you adjust the internal FX sends, for example, you find that the first 180° of travel of the rotary encoders takes you from -7dB to -18dB, which is quite a jump, whereas the last 180° gives you a much more sensitive control over the transition from here to 0dB - appropriate to an effect send, where you almost invariably want to maximise the level going to the effect, to give the best possible signal-to-noise ratio at the other end. Although rotary encoders, as used on the 328 and many other digital mixers, are essentially linear devices – part of their value as controllers is that they have no centre position – part of the secret to good design in a digital mixer’s user interface is making the controllers feel right.

Another respect in which the ‘feel’ of the controls has been carefully looked at is the low and high EQ bands behave when you turn the Shape knob to take the EQ from parametric to shelving mode. When you do this, you’ll find that the sound changes smoothly, and the centre/cutoff frequency as displayed on the mixer’s LCD, does not change – this bends the rules on how the cutoff frequency of a shelving filter is defined, but it feels right. And since you should mix with your ears, not by looking at a display, we think the world of audio will forgive us...

WORDCLOCK

Wordclock is the glue at the heart of digital systems, and understanding a little about it will avoid the frustration of not being able to solve some basic and simple problems. Symptoms of wordclock-related problems include unwanted pops or clicks, and a metallic edge or other distortion on digital sources.

In any situation where several digital audio devices are connected via their digital rather than their analogue audio connections, all the devices must be wordclock synchronised to avoid data transfer problems. This means not only that the different devices send and receive their data at the same sample rate – 48kHz, for example – but that their internal clocks run precisely in sync. This ensures that all units send, receive and process their data streams in a precisely simultaneous manner, avoiding a major source of distortion and poor audio quality. To understand how this distortion arises, consider a simple situation in which a DAT recorder's digital output feeds a mixer's digital input. They are both using a 44.1kHz sampling rate, but their clocks are not synchronised. The mixer's input should have 44,100 'slots' every second that have to be filled by samples, and the DAT recorder should be sending it 44,100 samples every second. But if there is only a slight drift in one of the clocks, some of those samples will be missed or will 'jump' a place, and that means distortion.

So, in any system that sends digital audio between several different devices, one unit should be the wordclock 'master', and the others should be 'slaves'. Setting this up is usually very easy, as the slaves can generally derive their wordclock timing from their digital audio inputs – SPDIF, AES/EBU, ADAT or whatever. Wordclock can also be carried via a separate connection. In either case, the wordclock signal is present whether audio is sent or not.

A. Choosing a Wordclock Master

If you are using several digital connections at once, you may not be left with much choice as to which device should be your wordclock master. If, for example, you are using a sampler with a digital output, you may find that it does not have any facility to slave to an external wordclock input, and must therefore be the wordclock master. On the other hand, any digital recorder in the system should be the master, though some recorders are quite happy to slave to external wordclock. As a rule, PCI cards installed in computers behave best when they are wordclock masters rather than slave. Trial and error will tell you which master/slave combinations works best.

B. Where Does Timecode Fit In?

Timecode synchronisation is different to wordclock synchronisation, but equally important in another way. While wordclock enables digital audio to be transferred without problems, by keeping the various audio inputs and outputs running at the same frequency, timecode enables recording and playback devices to run in sync with one another, and with sequencers and desk automation. Timecode is a signal that carries a regular series of absolute time values (hrs:mins:secs:frames). There are different frame rates (hinting at timecode's origins in the need for a means to synchronise film equipment), of 24, 25, 30 (drop and non-drop) frames per second. As in wordclock, one device is defined as the master, and its timecode keeps every other recorder running along in synchronisation. The two timing systems are quite independent, and a recorder that is the wordclock master in a system could be the timecode slave of another device (the 328, for example, which can generate all timecode frame rates).

C. Using Tape & Group Outputs

There are two approaches to using outputs to tape. The first is to use your mixer's group facilities to combine several channels together for recording – the outputs for the eight groups appear at the 8 TRK A outputs on TDIF and ADAT optical outputs. The second is to route channels directly to tape. (In an analogue mixer, this has the advantage of a cleaner signal path, though in a digital mixer like the 328, you can route through groups without adding any noise or distortion.)

The first approach is suitable for recording several instruments together onto one or two tracks – a multi-miked drum kit, for example. This makes sense when you don't have enough tracks to record every instrument to its own track. The second approach is suitable where you need to record more than eight tracks at once, perhaps in a live situation, and is simpler. Channel 1 always feeds track 1, channel 2 feeds track 2, and so on up to 16.

When you play back your recorded tracks, they come in on channels 17-32. Although they're called Tape Return channels, they have exactly the same facilities as channels 1-16.

D. Recording Instruments

Let's assume that you want to put together a track that uses live vocals and guitar with programmed keyboards and drums. You've programmed the basic song structure on your sequencer, and you're using two DA88s as above for your recording medium.

The first thing you'll need is a guide for everyone to play along to. As you already have drums and keyboards programmed, you could run your sequencer in sync with the tape transports as you record vocals, guitar and percussion. Alternatively, you could record a rough stereo mix of keyboards and drums to two tape tracks, which means that in future sessions you can get your guide tracks up without having to worry about loading the right

samples into your sampler, or selecting the right keyboard sounds.

E. Electric Guitars

You shouldn't plug electric guitars or bass guitars straight into a mixer. If you want to record a clean guitar sound, then use a DI box to get a good level into the desk. Some DI boxes require phantom power, and should be connected to the channel's XLR inputs (make sure the 328's +48V phantom power is switched on). If it's a distorted guitar sound you're after, you can either mic up the guitar cabinet, pointing a mic right at the speaker cone, or use a multi-effects processor. If you have enough tracks to take advantage of it, you can record the multi-effects in stereo – the processors almost invariably have stereo outputs, and flanging, chorus and delay effects sound better in stereo. See below for tips on using stereo channels.

Try applying compression while recording guitar tracks to keep levels under control. Using the compressor/gate algorithm on the 328 will allow you to gate out the unwanted buzz and hum that tends to come with distorted guitar settings, and which you'll only have to remove before mixing anyway.

F. Recording Vocals

The Spirit Digital 328 has high-quality mic preamps on channels 1-16, which are vital to recording good vocals. A good condenser mic will generally give the best results, as they are more sensitive and revealing than dynamic mics. Condensers require +48V phantom power, which the 328 can provide via the XLR channel input sockets. If you have the luxury of a choice of mics, be open minded about what to use – try them all before deciding what to use.

Set your mic up around 9 inches from your singer, with a pop shield (nothing really works better than the old stocking-and-coathanger trick) between singer and mic to avoid problems with

'plosive' sounds. Remember that making your singer as comfortable as possible will help to get a better vocal performance, so be sensitive to their needs. You may find that the 100Hz high-pass filter is useful to cut out rumble, though this is generally more of a problem live than in the studio. Avoid EQ'ing vocals when recording, as it will tend to make them sound artificial, and you'll regret it later.

Set up a vocal monitor mix on headphones using one of the Aux sends – a quick way to do this for, say, Aux 1, is to hit the Aux 1 button above the E-strip, so that the 16 rotary controls let you set Aux 1 send levels for channels 1-16, then hit Aux 1 again to set the send levels for channels 17-32.

Because vocal levels tend to vary widely, it's advisable to use compression to level things out. Try using one of the internal dynamics processors, using the compressor algorithm (with a 10ms attack, around half a second release, and ratio of 4:1 and 8:1). Alternatively you could use a favourite outboard compressor, plugged into the channel insert point.

G. Keyboards and Stereo Recording

Although one of the advantages of using MIDI is that you can record keyboards and samplers on a sequencer, and play them 'as live' into the mix without ever going down to tape or disk, you may instead want to record keyboard parts to multi-track. One reason is that it forces you to commit to a certain sound or version, rather than leave everything in a half-finished state. Also, it means that if you plan to take your project from your studio into a larger room for final mixing, you can take ADAT or DA88 tapes, or a computer with all your tracks recorded on HD, rather than a rack of keyboards.

But whether you do this, or leave the electronic instruments to be played via MIDI at mixdown, you'll want to use stereo inputs on the Digital 328

to get the benefit of all those stereo samples and stereo outputs. There are two stereo input channels on the 328, with all the the EQ, Aux and FX facilities of the mono channels. You can go directly into these via stereo digital interfaces – either SPDIF or AES/EBU – or use analogue inputs as on the mono channels. Remember that if you use the digital inputs you should take care with word-clock, as an unsynced audio source will show distortion of some kind, which will negate the benefits of using the digital inputs. If you have more than two stereo sources, you can link adjacent mono channels to behave as another stereo channel, with all EQ and Aux settings shared between channels.

H. Mixing Down

When you've recorded everything you need for your track, it's time to mix everything down to a stereo master recording prior to CD mastering, or playback on any other stereo medium. Whether you're using a sequencer or not, the 328's automation features make this much easier than with a non-automated desk.

The first step when preparing your mix is to set the desk in a neutral state. For all tape returns, check that EQ is off, and all sends are set to 0. If you are working in a MIDI-based studio, you may have used EQ and effects quite heavily on some of the synths and samplers, and now is the time to rethink what is essential to the sound, and what is cluttering up the mix and should be removed before proceeding. Mute all inputs that aren't being used, and for any analogue inputs, set the gain as high as possible without clipping – use the channel metering to check this. You can now start working up a mix.

Begin with the basics of the track – probably bass and drums. Set their relative levels so that neither dominates the other, and keep them in the centre of the stereo image. Use the eight internal groups to group together instruments that you will want to fade up and down together, such as

backing vocal parts, drum and percussion parts, string arrangements and so on.

Try using EQ to boost each instrument in a slightly different band, preferably the frequency range that carries most of the character of that instrument. This will help to distinguish them from one another, and make them sit apart in the mix. Using a little reverb on each channel tends to bring everything back together, creating the impression that the various sounds are all playing in the same acoustic space.

I. Using Effects

With two Lexicon effects processors inside the 328, it's easy to add a wide range of high quality studio effects. As well as a full selection of classic studio reverb and delay-based algorithms, each processor has more distinctive off-the-wall effects such as Resonant that are worth trying out. When setting up your effects mix, remember that you'll get the best signal-to-noise ratio, and therefore the most professional sounding mix, by keeping the send levels high and the return levels low.

You can also use external processors, using one of the Aux sends to take a signal to the processor, and bring the effect back on stereo inputs or linked stereo channels. If your processor has digital inputs and outputs, by assigning an Aux send and stereo return to the 328's stereo digital inputs and outputs, you can keep the whole effect send/return loop in the digital domain.

J. Using Dynamics

As well as the two internal Lexicons, the 328 also has two internal digital dynamics processors. These let you apply compression and limiting while track-laying, to avoid clipping and distortion and get a good level down to your recording medium. Gating any mic'd sources will keep unwanted background noise out of your mix. Because you can select the source for each dynamics processor (the source that determines gain change) independently of the destination (the channel or chan-

nels whose levels are changed), you can get creative with the dynamics section. Try using a gate, keyed from a rhythmic sound such as a hi-hat, to modulate the volume of a continuous sound such as a keyboard pad.

K. Using Automation

Once you have set up a basic mix – instruments sitting in the right place in the mix, the right effects balance – think about where you want to change this through the mix. By using snapshot automation, running against timecode, you can change fader levels, effects and aux sends, EQ and bus routing, wherever you need. Store the variations on the basic mix in different snapshot locations, and give them names if you like. Now give the snapshots timecode locations that correspond to the point in the song where you want the change to happen. (Stop the song at whatever point the change needs to happen, such as when a solo starts, or an instrument drops out and you want to mute its track, and give the snapshot this timecode value.) If you're anticipating a solo, set the snapshot to change just a little ahead of the first note that it will affect, in order that you don't cut off the start of anything – and if you're muting after an instrument has finished playing, be careful not to mute before the last note has died away completely.

If you're using a MIDI sequencer, then you can use dynamic automation and moving faders to mix in a different way. If you are using MIDI to play instruments as well, and you have a multi-port MIDI interface, it's good practice to dedicate one MIDI port to the automation data rather than mix note and automation data. By recording progressive fader and other changes, you can set up perfect fades at the end of your song, or to take instrument parts up and down. Remember that you can also automate EQ and other changes – in fact, almost every parameter on the desk, apart from the analogue controls, such as control room levels, can be automated and therefore used to create effect in a mix.

Use automation to give your track the dynamics that it needs. Try having some instruments that sit back in the mix for much of the time, then sit up – higher volume, less effect – at key moments. But remember above all that it's your track, and you're the best judge of how it should sound. A digital mixer such as the 328 gives you an amazing degree of production power, and it's up to you to decide how to use it.

L. Stereo Outputs

When you master your recording, you can record to both digital and analogue media – balanced analogue outputs are available alongside unbalanced phonos. If you're using the digital outputs to DAT, MD or an MO recorder, use the highest resolution available to you. If your media allows it, make your recording at 20 or 24 bits rather than 16 – both the SPDIF and AES/EBU outputs can be set to run at any of these wordlengths.

328 FOCUS: MIXER MAPS

In a studio based around a computer and PCI card, you can extend the integration between sequencer and the 328 beyond audio. Cubase Audio VST, Emagic Logic, and Cakewalk all allow custom mixer maps (on-screen MIDI faders) to control and be controlled by the 328's dynamic automation. By integrating sequencing and mixing to this degree, you have still more flexibility in your writing and mixing techniques. Mixer maps can be downloaded free of charge from the Spirit web site.

328 FOCUS: LINKING CONSOLES

If you're working on a project that grows and grows, you can cascade two 328s via a special digital link, to create what is in effect a single automated digital mixer with 32 moving faders and 84 inputs on mixdown.

DIGITAL 328 FAQs

How does the reliability of the Digital 328 compare to an analogue console? Traditionally, analogue consoles have been constructed using a single printed circuit board for each channel, plus one board for each function in the master section. Each circuit board uses conventional electronic components, and the boards are linked together by ribbon cables. The Spirit Digital 328, on the other hand, uses 'surface mount' technology and high density printed circuit boards. Surface-mount technology has been widely used in the computer industry, improving reliability and speed.

IS THE OPERATING SYSTEM UPGRADEABLE?

Yes: newer versions of the 328 Operating System will be made available through the Digital 328 web site (www.digital328.com), or as an upgrade kit from Spirit distributors. These updates will be accessible as soon as new versions are released, and (internet updates only) available at any time of the day. All software upgrades are provided free of charge by Spirit.

WHAT DIGITAL INPUTS AND OUTPUTS ARE FITTED AS STANDARD?

- * 2x8 track inputs Alesis ADAT optical format
- * 2x8 track outputs Alesis ADAT optical format
- * 2x8 track inputs TDIF format
- * 2x8 track outputs TDIF format
- * 1x8 track variable outputs Alesis ADAT optical format
- * 1 AES input on XLR
- * 1 AES output on XLR
- * 1 S/PDIF input on RCA
- * 1 S/PDIF output on RCA
- * 1 MIDI In 1 MIDI Out 1 MIDI Thru
- * Wordclock input on BNC
- * Wordclock output on BNC (TTL level)
- * 1 SMPTE input on quarter-inch jack
- * Sony 9 pin for Machine Control (RS-422)
- * Cascade Port on 26 way HD D-type connector

HOW MANY EXTERNAL DIGITAL RECORDERS CAN BE USED AT ONCE?

Two. The Spirit Digital 328 has two Alesis Optical and two Tascam TDIF interfaces located on the back panel, one of each for each set of eight tape inputs (17-24, 25-32). The front panel includes transport controls, with locator points, and the ability to arm (and indicate arming) any of the 16 tracks for recording. If two 328s are linked via the cascade facility, up to four recorders (with 32 tape channels) can be used.

ARE THERE ANY DIGITAL AUX SENDS?

Yes. A third Alesis ADAT format Optical output (labelled Aux) is located on the back panel. The source for the eight channels of this AUX Optical Output can be set to come from the AUX bus, FX bus, MIX output or Group buses. Companies such as Alesis and TC Electronics are already designing effects units with Optical In/Out, as well as standard quarter-inch jacks.

CAN A CD OR DAT MACHINE BE CONNECTED DIGITALLY?

Yes. Most professional, and some consumer, audio DAT and CD-R machines include stereo digital input ports. Simply assign the MIX Outputs to the digital AES/EBU or SPDIF outputs, and send that signal in the digital domain to the Input of the DAT or CD-R recorder.

WHAT TYPE OF MIC PREAMPS DOES THE DIGITAL 328 HAVE?

The Digital 328 incorporates Spirit's own UK-designed-and-built UltraMic Plus⁺ mic preamps on channels 1-16. This acclaimed preamp design has the ability to handle a wide range of signals, with a massive 66dB of gain range and +28dBu maximum input level. This caters for virtually all high level outputs that are likely to be passed through the preamp.

HOW DO GROUP ASSIGNMENTS WORK?

Simply select the channel you wish to assign, then choose the group(s) to which you wish to assign

them by pressing the button corresponding to that group in the Select panel on the right side of the console. Alternatively, you can press and hold any one of the Group assign buttons and select the channel(s) you wish to assign to that Group Out. You can also query which channels are assigned to specific Group Outs. Simply press and hold one of the Group assign buttons to see instantly which channel(s) are assigned to a specific Group at any time.

WHERE DO THE GROUP OUTPUTS ROUTE TO?

You can assign the Group Outputs the ADAT optical outputs or the TDIF connectors, or to the additional optical output (this is the how the Groups are routed on power-up). The groups can also be routed to the Mix by holding down the Route to Mix button and selecting the Group Select buttons

ARE ROUTING AND BUS ASSIGNMENTS STORED IN SNAPSHOTS?

Yes; every digital function within a channel, such as EQ, Pan and Group Output assignments can be stored internally using the Snapshot function.

WHAT ARE THE EQ FREQUENCY RANGES?

40Hz to 800Hz for the low frequency band, 200Hz to 8kHz for the mid band, and 1kHz to 20kHz for the high frequency band.

HOW DOES THE SOLO/PFL SYSTEM WORK?

Each channel has its own Solo button - press once to solo the channel, and again to clear it. In the Solo Control panel you can switch globally between PFL (Pre Fade Listen) and AFL (After Fade Listen), and select SIP (Solo-In-Place). There is a Clear Solo function which will clear the solo from all 16 channels. There is also also a trim adjust for the Solo system, which is the final fader on the masters bank.

HOW DO I SET THE AUXES TO BE PRE-/POST- FADER?

Hold down the Aux/FX Pre switch located on the right hand side of the console, and select which of the Aux and FX sends you want to be globally Pre/Post fader by using the Aux Select switches located in the Rotary Controls section.

WHICH PARAMETERS CAN BE AUTOMATED?

Snapshot Automation allows every function and parameter on the desk to be instantly recalled, except those associated with analogue controls eg. TRIM, HP Filters, C/Rm + phones output level controls. Dynamic Automation allows for individual control of each function and parameter through MIDI. All non-analogue functions of the console - including level, pan, EQ, and dynamics, aux level, and effects parameters - have been assigned MIDI messages which allow software control through almost any MIDI sequencing package.

HOW DO THE UNDO AND REDO FUNCTIONS WORK?

Spirit Digital 328 offers non-destructive editing by using the Undo/Redo function. This is useful for comparing new EQ settings or FX settings. Whenever functions have been updated, you can press the Undo button to return to the original settings. To return to the new settings, press Redo. There is only one level of Undo/Redo to avoid confusion. Snapshots can be used if multiple levels of comparisons are required.

TROUBLESHOOTING

I CAN'T HEAR ONE OR MORE INPUT CHANNELS AT THE MIX OUTPUT, EVEN THOUGH THE FADER IS UP AND THE CHANNEL IS NOT MUTED...

Check that the channel is routed to the mix bus, either directly or via the groups. Hit the select button(s) for the channel in question, and see whether the Route To Mix button or any of the Route To Group buttons are lit. If it's routed to a group but you still can't hear it, check the group's level, and that it isn't muted. To do this, select the Master Fader bank, and check fader positions and mute buttons for the eight faders on the far left.

ONE OR MORE CHANNELS SOUND VERY ODD - I KNOW THE INSTRUMENT SHOULDN'T SOUND LIKE THAT.

First check whether you've applied radical EQ to the channel - press and hold the EQ In button in the Select panel, and see whether the select buttons on the channels in question are lit. If you're using two mono channels to record a stereo instrument, check that both channels are in phase, and that the channels are panned left and right. (If you have copied settings from one channel to the other, to give them the same EQ and FX/Aux settings, then you may have copied the pan settings as well. In the Chan Copy Setup menu you can disable the copying of pan settings between channels.)

I'M HEARING UNEXPECTED POPS AND CLICKS, OR DISTORTION, ON DIGITAL INPUTS.

This is almost certainly a wordclock problem. If you are using an ADAT with your 328, and using the optical interface, check that the ADAT is set to receive wordclock. Ensure

that wordclock is being properly distributed, and where appropriate that all devices are set to slave to timecode from the appropriate inputs. Try making a different device the master - some DAT recorders, for example, must be the wordclock master when or changing the clock frequency of the wordclock master. Try turning everything off, and turning them on again in the order they are connected in the wordclock chain, starting with the master.

I CAN'T WORK OUT HOW TO ROUTE CHANNELS IN STEREO THROUGH GROUPS TO THE MIX.

In order to use groups in stereo, you must first link adjacent pairs - 1&2, 3&4, 5&6, or 7&8. If groups are not linked in this way, they are automatically routed in mono to the Mix. (Note that this applies to 328 software V1.1 and later). To link two groups, press and hold the Group Link button in the bottom right hand corner of the front panel, and press the select button for the odd-numbered Group. All channel pan settings are now preserved when you route channels through linked groups to the Mix. To route the groups to the Mix, press and hold the Route To Mix button, and press either of the select buttons for the paired Groups.

MY SEQUENCER ISN'T RECORDING MY FADER MOVES AND OTHER CONTROL MOVES.

Check how the sequencer is set up - it is probably not recording MIDI controller data, which is used for the 328's dynamic automation. Change your recording preferences to allow this to be recorded. Make sure MIDI Automation is switched on in the 328 Automation Setup menu.

WHEN I USE DYNAMIC AUTOMATION, THE FADERS ARE JUMPING AROUND INSTEAD OF MOVING SMOOTHLY.

This is caused by a MIDI loop via your sequencer. Turn off MIDI Thru so that data from the 328 is not re-sent to its MIDI input.

FURTHER INFORMATION

For further information please contact us at the details overleaf, or alternatively visit our website:

www.digital328.com

You will find information on the brochure, user guide, interface options, mixer maps, latest software and much more.

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