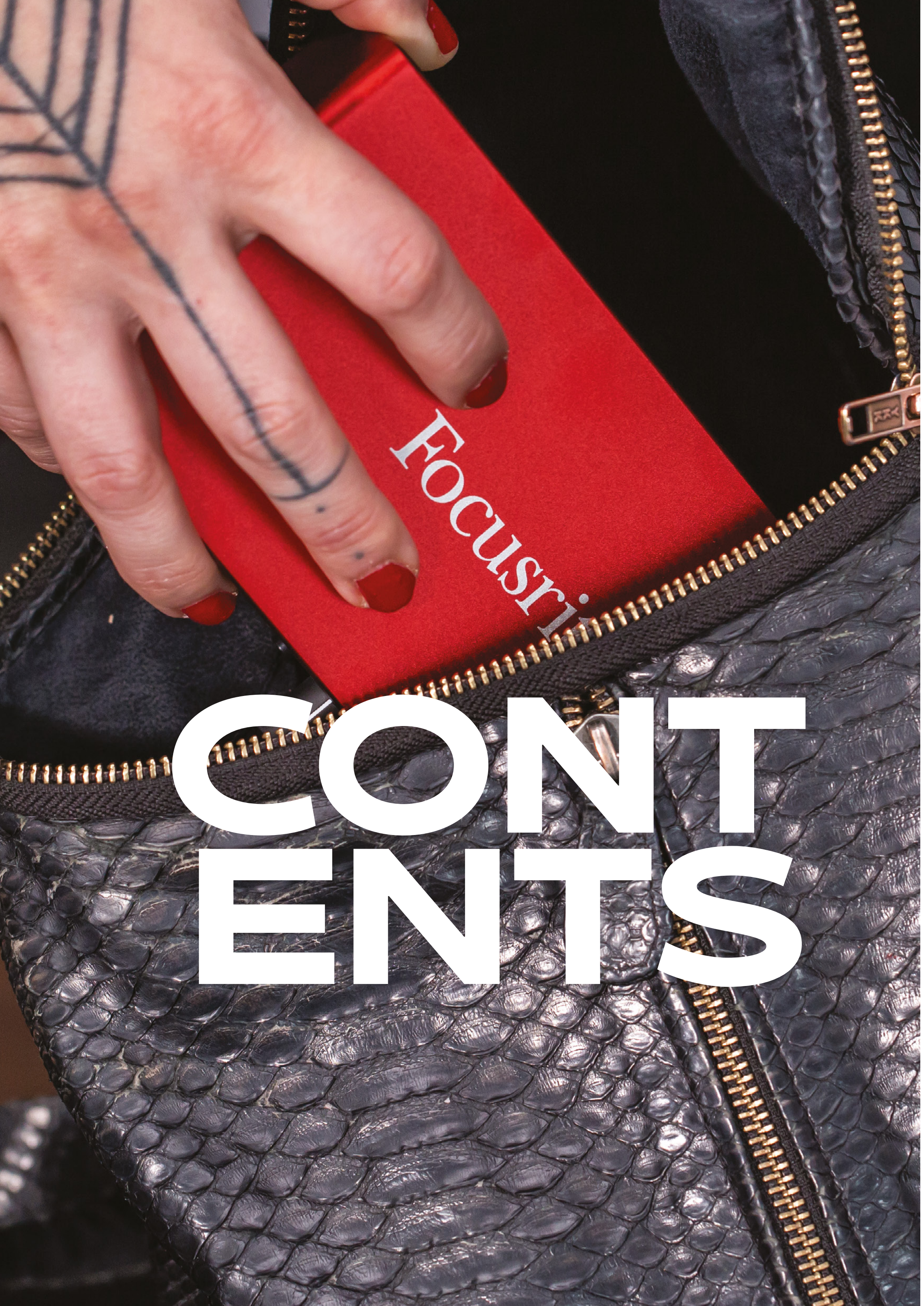
THE INTERFACE USER'S GUIDE TO MAKING MUSIC

If you make music with a computer, your audio interface is the single most important item in your studio inventory. The interface ensures that signal passes to and from your various audio devices: microphones, speakers, synths, outboard equipment and in some cases, MIDI equipment. A good interface will do this day-in, day-out without disruption, and in so doing, enable you to be in the creative zone 100% of the time.

Focusrite®



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INTRODUCTION

Every audio interface performs two critical processes in a computer-based signal flow. Firstly, there's the audio and data processing and interfacing: managing the inputs and outputs from external and in-computer sources, amplifying analogue audio signals, balancing monitor outputs, and mixing digital and analogue audio, all while ensuring your audio is crystal-clear, phase-aligned and unsullied by these processes. Second is analogue-to-digital (A-D) conversion: precisely representing the analogue signals in digital information, and vice versa (D-A) converting your digital signals from DAWs and other software audio sources into clean, analogue signals.

But despite its importance, an interface should be invisible most of the time, yet provide enough flexibility so that you can quickly set up a headphone mix for a session player, or experiment with routing your audio around the hardware devices in your studio.

THE ANATOMY OF AN AUDIO INTERFACE

The front panel of your audio interface provides controls to adjust some of its features. Features without front-panel controls can be adjusted via control software.

Interfaces follow a left-to-right workflow, with input controls on the left and output controls towards the right. You'll find gain controls for each of the mic preamps, and input-type selectors, allowing you to choose from mic, line or instrument options. Other preamp controls include phantom power, gain-reduction pads (useful for taming extra-loud signals) and phase-reverse controls.

FRONT PANEL

As for output controls, the amount of options here varies from interface to interface. The most basic devices featuring a monitor level and headphones volume, as with the Focusrite Scarlett 2i2. Some devices have mute and dim buttons, which provide console-esque control for managing your monitors. Other buttons and switches on an interface's front panel will allow for manual selection of sample rate and clock source, which lets you quickly configure settings without having to dive into the control software.

Metering is another important front-panel feature, which helps you set your levels by glancing at the color of the display. Focusrite Scarlett interfaces feature gain halos, which glow green, amber or red. Green shows signal present, amber indicates that the preamp is almost clipping, and red shows clipping, which should be avoided.

Status LEDs are the final thing you'll commonly find on an interface: depending on your interface's capabilities, separate 'connected' and 'locked' lamps show when you have a stable connection, and when you have sample-rate lock.

COMBO INPUTS

THESE MULTI-PURPOSE INPUT PORTS ACCEPT QUARTER-INCH AND XLR CONNECTORS

METERS

GAIN HALOS INDICATE THE LEVEL OF THE INPUT SIGNAL. GREEN MEANS GOOD, AMBER MEANS STRONG, RED MEANS THAT THE CONVERTERS ARE CLIPPING (BAD!)

INSTRUMENT INPUTS

HIGH-IMPEDANCE CIRCUITRY PROVIDES A BETTER MATCH FOR INSTRUMENTS SUCH AS GUITARS AND BASSES.

AIR

ADDS HIGH-FREQUENCY INFORMATION THAT BOOSTS THE PRESENCE OF VOCALS, GUITARS AND OTHER ACOUSTIC INSTRUMENTS

HEADPHONES

EACH HEADPHONE SOCKET HAS A DEDICATED LEVEL CONTROL. CUSTOM HEADPHONE MIXES CAN BE MADE IN SOFTWARE.

INDICATOR LEDS

THESE SHOW THE STATUS OF VARIOUS AUDIO INTERFACE FUNCTIONS. THE USB SYMBOL ILLUMINATES WHEN A SUCCESSFUL CONNECTION IS MADE WITH THE HOST COMPUTER.

METAL BOX

A ROBUST HOUSING NOT ONLY KEEPS YOUR INTERFACE SAFE ON THE ROAD, IT REDUCES RF INTERFERENCE.

MONITOR CONTROL

THE 'BIG KNOB' ADJUSTS THE LEVEL OF THE STEREO MONITORS

GAIN CONTROLS

THE GAIN CONTROL INCREASES THE AMOUNT OF SIGNAL ENTERING THE MIC PREAMP. ITS IMPORTANT TO SET YOUR LEVELS BEFORE YOU START RECORDING.

PHANTOM POWER

THE PHANTOM POWER BUTTON ON A MIC PREAMP ENGAGES A +48-VOLT SUPPLY, WHICH TRAVELS DOWN THE MIC CABLE AND ACTIVATES THE INTERNAL CIRCUITRY OF A CONDENSER MICROPHONE.

PADS

ENGAGING THE PAD REDUCES THE INPUT GAIN, WHICH IS USEFUL WHEN YOUR INPUT SIGNAL IS VERY HIGH. PADS HELP WHEN USING SYNTHS AND DRUM MACHINES.

USING YOUR INTERFACE

THE BASICS

Large or small, all audio interfaces share some common features.

ANALOGUE INPUTS

Analogue audio inputs enable you to connect a variety of sound sources to your interface. Sources generally fall into one of three different categories: mics, instruments or line-level. A mic signal is low voltage and low impedance therefore needs to be boosted using a preamplifier because otherwise the signal will be too low to use.

An instrument signal has a similarly low voltage, but requires a high-impedance circuit to create an electrical balance. A line-level signal is stronger and usually does not need to be boosted before it enters the A-D converter. Many interfaces have Combo inputs that let you connect any kind of device, often into a singular socket that accepts XLR and quarter-inch jack plugs. These inputs usually correspond with a gain control on the front panel, allowing you to increase or decrease the signal level by hand. Line-level-only inputs often don't have a gain control, because they operate on a common reference level.

LAUREN MARTINEZ

Brooklyn, USA

Lauren has been making music for several years, and over that time she has adopted a hardware-focused approach. "I strongly prefer hardware to software," she says. "I need to feel the instrument I'm working with — it's just more fun that way."

Recording on a computer plays an important role when it comes to capturing her ideas, and Focusrite has been with her throughout her journey. When she comes to choosing an audio interface, she has several priorities: "I look for sound quality, affordability, portability and enough I/O for all my gear. I've always had Focusrite audio interfaces since the beginning, and it's the only one I keep buying because it suits my needs perfectly."

Lauren acknowledges that her ability to be creative is dependent on her state of mind, and she actively seeks positivity in her life. "Some people write their best stuff when they're feeling down or going through something emotional, and some people write their best stuff when they're feeling good. I'm definitely the latter. I always find I'm the most inspired when I'm feeling really good so when I'm experiencing a block, I exercise, or go out and connect with nature. Meditation also really helps. I always find my inspiration hiding somewhere in the silence."

"I ALWAYS FIND I'M THE MOST INSPIRED WHEN I'M FEELING REALLY GOOD. SO WHEN I'M EXPERIENCING A BLOCK, I EXERCISE, OR GO OUT AND CONNECT WITH NATURE"

PHANTOM POWER

Phantom power, commonly marked as '48V' on music-making equipment, is required by some microphones. When phantom power is applied, a voltage (48 Volts, or thereabouts) is sent down the XLR cable to the connected microphone, enabling its internal circuitry to operate. Note that, even with phantom on, the mic's signal will still need preamplification; the 48V supply is simply there to enable the mic to operate.

As every engineer will tell you, best practice is to connect all your mics with phantom power off, then only engage phantom power on mic channels that need it once they are connected. If your mic preamp does not have a dedicated 48V button, there may be a global phantom supply switch, or phantom might be switchable in banks (say, channels 1-4 and 5-8 on an eight-channel interface such as the 3rd Generation Focusrite Scarlett 18i20, pictured). Sometimes, phantom power can also be engaged using an interface's software control panel.

Lastly, you should be extremely cautious when using older equipment, especially vintage ribbon mics, which can be critically damaged if phantom power is engaged on its channel. Even modern gear can be disrupted by phantom power, so it's always best to check the documentation for your equipment before plugging it in.

ANALOGUE OUTPUTS

There are two types of analogue output found on most audio interfaces: mono line-level outputs and headphones outputs. Most are of the line-level variety, carrying mono signals that allow you to connect to studio monitors and outboard hardware, such as compressors, effects units, consoles and summing mixers.

Commonly, your interface will provide balanced outputs, although some entry level devices have unbalanced connections. At the higher end of the spectrum, some audio interfaces enable you to change the reference level of the outputs (commonly between -10dBV and +4dBu). This is important for interfacing between professional equipment like recording consoles, which like to see a higher level, and things like synths and low-budget effects, which often operate at lower levels. It may seem obvious to the seasoned studio user but, in most cases, only the stereo monitor outputs and headphones connections have level controls. (Although, again, some more esoteric devices let you tweak the level of every output.)

Why? Well, it's very useful to be able to turn up/down the level of your speakers and headphones during a session — to have a quick chat without being drowned out, or to pump up the volume to hear how the bass kicks, as examples. But in systems where connections are being made out to analogue outboard, it's crucial to maintain consistent levels.

SIGNAL TO NOISE RATIO & THD+N

Though all line-level outputs may seem equal, there are defining factors that can have an effect on the quality of the audio leaving your interface. These factors are signal to noise ratio (measured in dB), and the amount of noise and distortion in the signal, which is referred to as Total Harmonic Distortion and Noise, or THD+N for short. This figure is usually represented as a percentage, and the lower the number, the cleaner the signal.

As the name suggests, the signal to noise ratio is the difference between the highest possible signal and the level at which noise is present.

Generally speaking, the higher this figure, the greater dynamic range your interface will be capable of producing. That said, there are many other dependent factors on whether your interface will actually be able to take advantage of this range, so the signal to noise ratio should be considered a ‘best case scenario’ — a maximum performance figure for your audio, so to speak.

Meters provide a visual representation of your audio signals, often multi-coloured LEDs that illuminate differently depending on the strength of the signal. All but the most basic interfaces will provide some kind of metering functionality. Metering is important for making sure your levels are healthy: not too high where there's a risk of distortion; not too low where they sound weak. In the bygone days of recording on large analogue consoles to multichannel tape machines, it was a creative decision to push the levels into the red, forcing the circuitry to distort in a sonically attractive way.

Audio interfaces and recording on a DAW do not follow the same principles — when a signal reaches 'max' in the digital domain (or 0dBFS if you're counting) it flatlines, creating unpleasant distortion that can ruin an otherwise perfect take. So, the golden rule when recording in a DAW is to check the input level over the range of sounds that will be captured before you start recording. Ask your singer to bellow out his loudest phrase, and your guitarist to play her loudest lick.

One last tip: leave some headroom in your preamp, because musicians have a tendency to get louder once the blood starts pumping!

HEADROOM

Headroom is a term used to describe the amount of capability your audio signal has before distortion occurs. When recording signals where the input level is unpredictable (i.e. with any acoustic instruments and vocals; even synths and drum machines) it's always a good idea to 'leave some headroom' in the system. In real terms that means not maxing out the input levels, just in case a stray rim shot or extra-loud bass slap comes into a mic.

When recording at 24-bit or higher, good practice is to have your levels peak at -12dBFS, giving you 12dB of headroom. This will allow your signals to breathe and stay free of distortion, and give you options in the mix when you come to finish your projects.

COMPUTER CONNECTIONS

The computer connection is obviously the link through which the audio inputs and outputs reach your computer for processing. Though Firewire and Thunderbolt interfaces are still widely used, USB interfaces are most prevalent.

On modern computers, such as the latest generation of MacBook Pro, you'll only find USB Type-C sockets. The data which flows through these ports, however, is not just USB-classified data, rather a combination of different protocols such as HDMI for displays, plus Thunderbolt — and of course a power supply. Most new interfaces will have USB-C connections, so hooking up a new device, such as a Focusrite Scarlett 4i4, to a new computer with USB-C is straightforward with the appropriate cable. However, the old-style USB-A connection (the ubiquitous half-inch wide rectangular plug) still dominates, so you'll often find a USB-A to USB-C connector in the box.

USB 3.X

In recent years, USB 3.0, USB 3.1 and USB 3.2 have been introduced. They bring tremendous improvements in terms of raw data-transfer capabilities — with maximum theoretical data rates of 5Gbps (gigabits per second) for USB 3.0; 10Gbps for USB 3.1; and a whopping 20Gbps for USB 3.2 — which is fantastic for hard drives and other devices that use ‘bulk’ data transfers. But these performance increases have no practical benefit in interfaces with less than 32 inputs and outputs. This is due to the architecture of the host computer’s driver stack and its handling of USB audio, which is ‘isochronous’ data. The stack schedules data transfers to and from audio drivers at millisecond frame intervals which means that, no matter how fast the data moves over the USB bus, the driver defines the limit on minimum latency achievable.

A good analogy is to think of it as a drainpipe and a tennis ball. The tennis ball is the data and the width of the pipe signifies available bandwidth. With the drainpipe set at the same gradient, letting go of the ball at the top of the drainpipe will see it arrive at the bottom in a given amount of time. That’s your latency, the time it takes to go from end to end. USB 3.x offers a much wider pipe, which in terms of the analogy means that it could allow a greater number tennis balls (more channels of audio) to travel down the pipe. But the balls would not travel down the pipe any faster, because the gradient is the same. This is the same when comparing USB 3.x and USB 2.0 in terms of the way they transfer audio data.

This is not to say that the higher bandwidth offered by USB 3.x does not have added benefit in some situations, say if you wanted an interface with literally hundreds of audio inputs and outputs. However, for the largest channel-count USB audio interfaces, USB 2.0 provides more than enough bandwidth for delivering all the inputs and outputs you’d ever need.

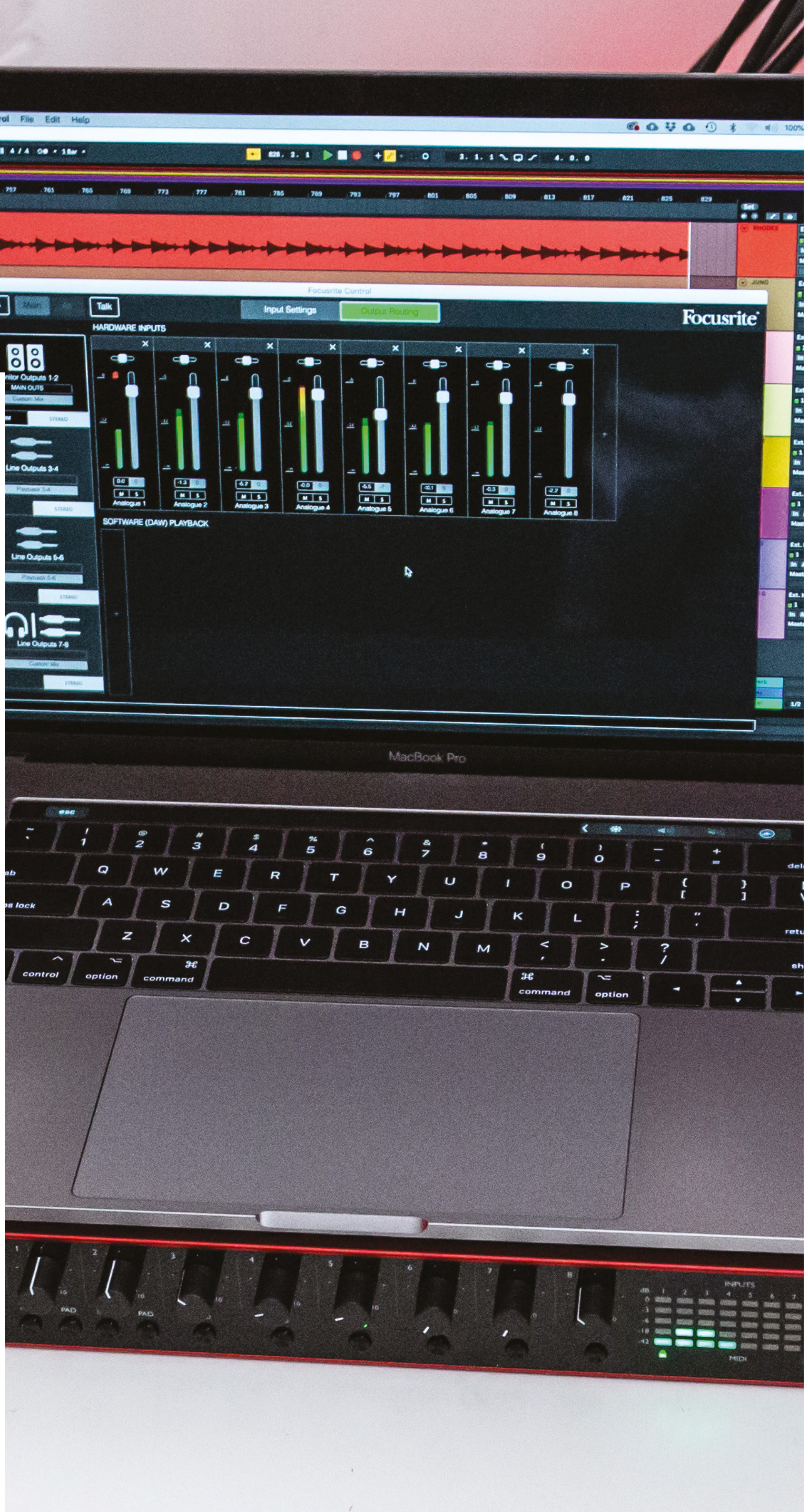
LATENCY

Latency is one of the most important things to consider when choosing an interface. Latency is the time delay that you experience between singing into a microphone (for example) and hearing your voice back once it has passed through the input stage of your audio interface, the A-D converters, the driver, your DAW, then back through the D-A converter, and out of your speakers or headphones.

At every stage, a slight delay is incurred, because of the calculations needed to process the audio or pass it on to the next step. In a low-latency environment, everyone is happy: the singer can hear himself in his headphones without a jarring delay; the engineer can apply plug-ins to her mix during tracking, and the whole band can monitor the session without needing to set up aux sends on an analogue mixer. Modern, high-spec interfaces , such as the Focusrite Clarett USB and Red ranges, can operate with near-zero latency figures, and this performance boost is often a big factor in the price point.

BUS POWER

Some audio interfaces can operate on the power supply that is provided down the USB cable: the same supply that can charge your phone or cordless headphones when you plug them into a USB port. USB 2.0 devices are permitted to draw up to 500mA at 5V from the host computer, while USB 3.x raises this to 900mA. However, these specifications refer to the maximum power that an attached device is allowed to demand: they don’t guarantee that the computer will be able to meet that demand. For this reason, to ensure the audio performance and integrity of the signal, larger audio interfaces will almost always have a separate DC power supply.



CLOCKING

Digital audio consists of a series of samples: measurements of the amplitude of a signal that are taken at regular intervals. In the case of CD-quality audio, there are 44,100 samples per second. Many professionals choose to record at higher sample rates, such as 96,000 samples per second, or 96kHz. Regardless of the sample rate in use, every digital audio device in a system needs to use the same timing reference or 'clock', otherwise audible glitches and errors will occur.

A single audio interface running on its own without any other digital equipment (aside from the connected computer) will happily run from its own internal clock. Likewise, there are no clocking issues involved in connecting a lone interface to other hardware in the analogue domain. However, when we digitally connect a second device, such as a mic preamp with a digital output, the topic of clocking has to be considered. The only way for digital audio to be transmitted successfully from one device to another is for the two devices to share the same clock signal. Or, using audio lingo, in any studio where devices are connected digitally, one of them needs to be the clock 'master', and all the others must be clock 'slaves', accepting their timing reference from the master clock.

The ways in which this is done vary depending on how many devices are being connected, and in what way. With ADAT and S/PDIF — the most common digital audio I/O formats on audio interfaces — clock signal is transmitted in parallel to the audio over the same cable, so when connecting digital devices to your interface using these formats, you can configure your interface to pick up the connected device's clock and slave from it. In the majority of interfaces, clock settings are found in the control software, and LEDs on the interface's front panel indicate clock status. On external devices with digital outputs, there's normally a cluster of controls that allow you to choose your sample rate and the status of the clock (master or slave).

WORD CLOCK

For expansive systems where numerous digital devices are connected to an audio interface, or when the connected device doesn't have a stable clock, a separate Word Clock setup is an option. Word Clock is only available on larger, higher-end audio interfaces, where it makes sense to provide greater flexibility for several digital machines to connect together. It requires an independent set of coaxial cables with locking BNC connectors, and careful attention to configuring and terminating the various devices in the clock loop. There are sometimes several ways to approach clocking on larger systems, so it's best to refer to the particular device's user guide for more setup advice.

DRIVERS

Whether on Mac or PC, all interfaces need a driver, with the exception of 'class-compliant' interfaces that use the default sound driver on the computer's operating system. The driver handles the flow of data between the computer and the audio interface, and has a huge impact on performance and stability of the device. A well-designed driver will allow low latency, solid reliability and will guarantee a stable connection to the host computer. A bad driver can turn a perfectly adequate interface into a lemon. It's good practice to keep your interface's driver up to date, keeping in mind the compatibility of your operating system and suchlike.

HANNIE

London, UK

Hannah and Annie produce and perform together as Hannie. "It's cool to play other people's songs and it's cool to be on tour with someone else," says Hannah. "But writing and producing yourself is so much more fun." Annie got her first Focusrite interface at age 13, and the duo have been using one ever since. "We use our Focusrite literally every day. We are able to record multiple instruments at the same time and make live recordings whenever we want."

"BEING ABLE TO WRITE AND PRODUCE YOUR OWN SONGS... THAT'S AWESOME!"

OPTIMISING YOUR STUDIO

Studios range in size and complexity, but whether you're recording a podcast in your bedroom studio or tracking bands in a dedicated recording studio, there are several tenets of good practice that can help to make the most of your gear and the space you work in.

ACOUSTICS

Room acoustics are a critical component of your setup. While we don't expect everyone to be making music in a professionally treated recording studio, there are some simple things you can do — often for free — to maximise your acoustic experience. Small, reflective rooms (i.e. with untreated walls and hard surfaces like windows and tiled floors) are the music maker's worst enemy, unless you're designing a reverb tank. When reflections from the room overpower the sound coming from your speakers, you can't make critical mix decisions or hear when low-end instruments like bass and kick drums are tonally aligned.

Domestic soft furnishings like rugs and curtains really help to dampen high-end reflections. For low-end absorption, bass traps can be employed to minimise the low-frequency standing waves, which create 'lumps' in the bass response of the room and make accurately analysing your bass a nightmare. A sofa or large cushioned chair will do a decent job of absorbing bass if proper acoustic products are out of the question. If you're in a bedroom, the bed will do a marvelous job of trapping the bass.

Remember that the objective with acoustic treatment is not to totally deaden the room — at that point, everything would sound boxy and flat. Some amount of 'liveness' is desired, and acousticians often recommend an absorption coverage rate of around 30% (where 70% of the whole surface of the room is left untreated).

If you are suffering from flutter echoes — high-frequency ping-pongs between hard surfaces — a diffuser will help to randomise the reflections. A well-placed book collection or stack of vinyl records makes a remarkably effective diffuser. You can even make your own with mismatched off-cuts of wood — there are plenty of YouTube videos to show you how.

SCOTT KAPELMAN (CONCLAVE)

New York, USA

Multi-instrumentalist and songwriter Scott is from New York, USA. He merges analogue and digital in his music making process: real instruments, digital hardware effects and computer-based processors and recording software. He uses a Focusrite Scarlett 2i2 to get audio into and out of his computer, either by plugging his instruments directly in and using software amp modeling and effects, or by using a multi-effects pedal and capturing the stereo output at line level.

"What I look for in an interface," says Scott, "is firstly clarity of sound and also ease of use. I use technology to flesh out demos and eventually record the songs that I write, so I need something quick and great-sounding."

Writer's block is part of the creative process for Scott. "If you hit hard times when creating, you need to find a way to push past those blockades. I keep on making stuff regardless of how it sounds. I can't wait around for inspiration to hit, if that was the case I'd be waiting for a while!"

"WHAT I LOOK FOR IN AN INTERFACE IS CLARITY OF SOUND AND EASE OF USE"

SPEAKER PLACEMENT

Loudspeakers act differently in different spaces and in different arrangements. In a small room, the sound pressure generated by the moving speaker cones will bounce off every surface, essentially turning the room into an extension of the speaker itself. Just because your speakers are expensive doesn't mean they will sound good in every room or in any configuration. Refer to your speakers' user manual for correct placement, but a general rule of thumb is to have them at the exact same level on the horizontal plane, pointed at at ear-height with each speaker approximately 30 degrees left and right of the centre line of your mix position. This will create a sweet spot where you can make critical decisions on your music. Speakers are normally designed to be oriented in one direction only — either upright or on their side — and straying away from these conventions will disrupt the way you hear the speakers and how authentically they will reproduce your music. Of course, great music has been made in sub-optimal conditions, but stick with convention and you'll have the best chances of making the right decisions when you come to mix your music.

In most signal chains, there are several phases of amplification, known as gain stages. If we take the example of a synthesizer plugged into an effects unit then into an audio interface, we can count a total of four gain stages: the synth output volume control, the

GAIN STRUCTURE

effects unit input, the effects unit output, and the audio interface input. At each stage, you have the option to boost the signal, but every time you do this, you risk adding unwanted noise and distortion to your signal. In our example, we might find that the synth output is too high for the effects unit, leading to input distortion which is carried into the effect and onwards through the signal path. To remedy this, you would trim the synth output to provide a lower signal to the FX. Every time the signal passes from one piece of equipment to another, the respective gain stages of each should be considered. When we connect the FX unit to an interface, we might find that level too low, even though the input to the effects unit is clean and undistorted. At this point, the output level of the effects unit can be increased to bring the signal up to a more manageable level for the interface.

Microphones, regardless of their type or performance, all need preamplification, a task usually performed by your audio interface's mic preamps. The mic preamps found on audio interfaces are designed to accept a wide range of mic types. A preamp's basic purpose is to add gain to the very low signal that comes from a microphone. The more gain your preamp is capable of supplying, the 'louder' you can make your signal. But with lots of gain can come signal distortion, noise and other unwanted, audible artefacts, especially when the gain is turned up all the way. It's for these reasons that you'll read about engineers preferring specific mic/preamp partnerships, and many studios using banks of esoteric outboard mic preamps to perform this specific duty.

A general rule of thumb with all gain issues is that you should avoid using too much gain at any one stage of the signal path — unless of course you are purposefully trying to distort the signal for creative effect, as you would with a guitar amp. The results will be audible in your recorded music: less noise from cranking the gain, less distortion from clipping and less post-production problem solving.

> **"SOMETIMES IT'S MORE IMPORTANT TO HAVE A VIBE THAN TO THINK ABOUT THE TINY DETAILS"**

KARUN

Nairobi, Kenya

Karun is an R&B artist from Nairobi, Kenya. She has been making music since the age of eight, having grown up in and around her uncle's recording studio. But being able to make music whenever creativity strikes is important. One of her first EPs was the result of her and some fellow music makers hanging out and recording in a friend's bedroom with a Focusrite Scarlett interface. "That brought the studio to our house, so we could make music whenever we got inspired."

One of her production mantras comes from attending a masterclass, where a mix engineer was breaking down a finished track by playing the individual parts. "There were layers and layers of vocals, and the engineer played the individual parts. Some of the parts were just 'ok', but they sounded amazing with the entire track playing!"

From that experience, Karun learned not to focus on the individual tracks, but rather to consider them as part of the bigger sonic picture. "Sometimes it's more important to have a vibe than to think about the tiny details. You don't have to be so particular."

CABLES

Cable-related issues are at the heart of many troubleshooting operations in the studio environment. Whether it's a faulty cable, a cable of the wrong type, or simply that you don't have enough to connect everything you want, cables often catch us out in the studio. It's wise to consider cabling as an integral part of your music-making system and, as you would with your computer or drum machine, ensure your cables are of good quality and are well maintained.

Generally speaking, there are two types of cable used for analogue audio. 'Balanced' cables and connectors are standard on a lot of audio equipment. The idea behind balancing is that, as well as a ground or screen connection, the wire carrying the signal is run alongside a second wire that will pick up the same interference. (Note that these various wires we refer to are encapsulated within a single cable housing.) The destination device then 'sees' not the absolute voltage in the signal wire, but the difference between the voltages of these two wires, meaning the interference is cancelled out and we are left with just the wanted signal. Creating a balanced connection requires a connector with three electrical contacts rather than just two.

An 'unbalanced' connection only relies on two points of electrical contact. Accordingly, unbalanced cable only has two wires: a ground and a signal. In an unbalanced connection, the ground wire performs two duties: it shields the signal wire from interference, and also carries part of the audio signal. Unbalanced connections are prone to picking up interference, and should be kept short to avoid noise in the signal.

On a side note about cables, passive loudspeakers (i.e. speakers that require amplification from an external power amp) also use a two-core cable, but because the signal traveling to it has different electrical parameters, a dedicated speaker cable is used, which has much a wider gauge to handle the higher voltage and current.

If you have a lot of analogue audio cables, it's a good idea to get to know how they are assembled, so you can fix problems as they arise. Making a few audio cables is a good way to obtain a greater understanding about the electrical signals that make the music happen. You'll need a soldering iron, some solder, a damp sponge and a 'solder sucker'. *[And a well-ventilated space! — Ed]*. Soldering is an art form in itself, but you'll find plenty of advice on the Internet to learn the basics.

Data cables are harder (if not impossible) to fix, because there are lots of connections in a very small physical space, and one cannot easily whip out the soldering iron and reconnect a loose end. That said, data cables are often more robust because their moulded connectors and built-in strain relief protects the internal points of electrical connection during transit and repeated plugging in and out. But data cables can still trip you up. You may have an old USB 1.1 cable in your stash — this was designed when bandwidths were meagre, and will not work favorably with the new USB 2.0 or USB 3.x device you're trying to connect. There's usually a clue to the spec of the cable written along the side of it.

CONNECTORS

Two main types of connector are used to terminate the cables that transfer analogue signals. One is the quarter-inch (6.35mm) 'jack'. Cables of this form have identical plugs at either end, and are reversible; the devices they connect have identical sockets on input and output. Jack connectors and cables can be balanced or unbalanced, and you can tell which is which by inspecting the plug. The additional wire in a balanced connection is linked to an extra ring on the barrel of the plug. It is therefore called a TRS connector, for 'tip-ring-sleeve', while the unbalanced version is just a TS. Depending on how balancing is implemented, using an unbalanced cable to join a balanced input and output may work, as will the opposite arrangement, but you'll only get the benefits of balancing if all three terminals are connected.

The other type of connector in universal use is the XLR. Unlike the jack, this is a 'gendered*' connector, and XLR cables have a 'male' connector at one end and a 'female' one at the other. The male connector has a circular barrel enclosing two or more pins, which correspond with sockets in the female connector to make an electrical connection. XLRs are available in many different configurations, but the only one widely used in audio is the three-pin version, which is almost invariably used to create a balanced connection. A female XLR plug is connected to the output of the source (a microphone, for example), with a male XLR plug connected to the input of the receiving device — a mic preamp, say.

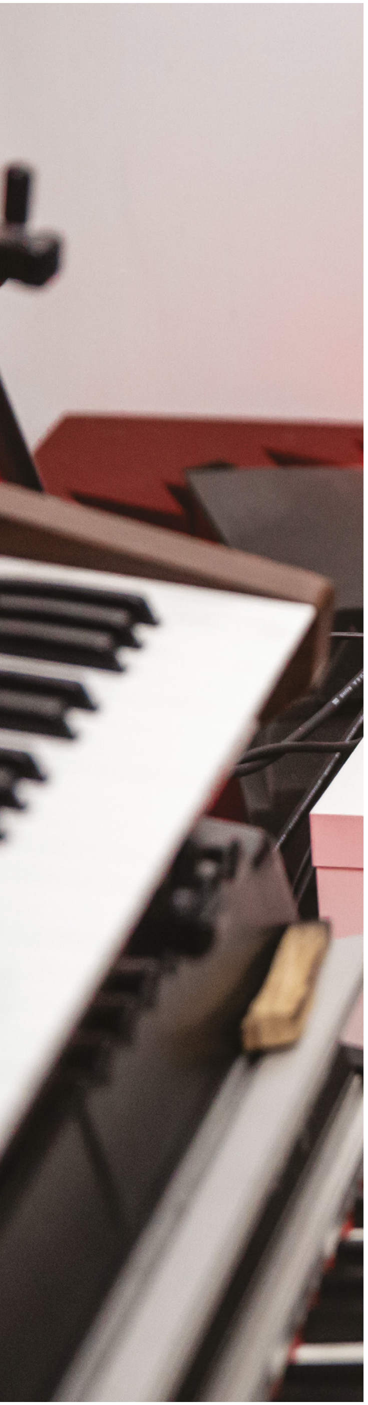
Three-way connectors can also be wired to carry two separate unbalanced signals. The quarter-inch and 3.5mm TRS jacks used on headphones, for example, carry one side of the stereo channel on the tip and the other on the ring, with the sleeve providing a common ground reference. Some audio interfaces and mixers also offer 'insert points' where a TRS jack combines an unbalanced output and input on the tip and ring.

You may also encounter a few other types of connector in use for analogue signals. RCA phonos are inherently unbalanced, and are mainly used on hi-fi equipment. And where many channels of audio need to be fitted into a small space, specialist multi-way connectors such as D-Subs and EDACs are employed. XLRs and RCA phonos are also sometimes used to transfer digital signals, although these are normally carefully labelled 'AES3' or 'S/PDIF' to avoid confusion with their analogue brothers and sisters.

Finally, analogue inputs on audio interfaces sometimes use the so-called 'Combo' connector, which incorporates a TRS jack socket in the centre of a female three-pin XLR. These are usually configured so that the TRS socket accepts line-level signals while the XLR offers a connection point for mics; in some cases the TRS can also be switched into a high-impedance mode suitable for plugging in guitars and other instruments.

**At Focusrite, we feel uncomfortable with this terminology. But we also acknowledge that these terms have historical precedent and have become standard in the industry. After much internal consideration, we chose to keep the terms 'gender', 'male' and 'female' in this context, only because no good alternative exists. If you feel strongly about this issue, we'd love to hear from you. editor@focusrite.com*

RECORDING 101S



CHOOSING A MICROPHONE

There are no hard and fast rules to choosing a microphone. Importantly, the price tag of a mic does not necessarily determine whether it will sound good on your sound source or not. (Dave Grohl has famously used a $100 Shure SM58 for many Foo Fighters records.) So take this advice with the understanding that you don't need thousands of dollars worth of microphones, or indeed lots of mics of different types, to make a record. You might just find a glittery karaoke mic at the junk shop that's perfect for you!

There are several different types of microphone found in a recording studio, and audio interfaces are designed to work with them all. Microphones, regardless of their classification, convert sound waves into electrical signals, but they do so using different methods. Dynamic mics operate just like a reversed speaker cone, where sound waves move an electromagnetic coil which generates a signal. Condenser mics obtain their signal by measuring the varying capacitance between two charged plates, which vibrate as sound waves pass them. Ribbon mics contain a conductive filament, which vibrates within the field of an electromagnet. Tube microphones are a type of condenser, which uses a vacuum tube in the internal electronics. These require a special power supply, which connects to the mic using a multi-pin connector (not an XLR). The XLR output for a tube mic is normally on the power supply. Regardless of type of mic, they all need to be plugged into a mic preamp to boost the tiny voltage from the transducer into something useful.

While studio mics operate within roughly the same parameters, they have different characteristics that govern the way they respond to sound. Some have a stronger input sensitivity and output levels than others, which makes them suitable for different applications. (A microphone with a sensitive input and very 'hot' output is not a good choice on a trumpet or a snare drum, for example, because it'll overload too quickly. Conversely, a mic that's good for a loud sound such as a kick drum or a guitar amp turned up to 11 might not be able to capture the delicacies of a harp or dulcimer.)

Generally speaking, dynamic mics have a low sensitivity and therefore can handle very loud and transient sources like drums. They usually also pick up less ambient noise and spill than condensers, and as such are very useful when miking instruments in close proximity to each other. They're also very useful in live situations where you need to manage the amount of spill making its way into the PA system. As a result, you'll often see dynamic mics like the Shure SM57 and Sennheiser MD421 used on drum kits and brass sections.

A condenser mic can usually handle just as much SPL (Sound Pressure Level) as a dynamic mic, but it will also pick up a lot of the ambient sound from the space you're recording in. This makes a condenser perfect for capturing the sound of a whole drum kit — it's conventional to use condenser mics as drum overheads for this reason.

POLAR PATTERN

THE POLAR PATTERN IS OFTEN DISPLAYED ON THE MICROPHONE. HERE, THE CARDIOID SYMBOL SHOWS THE FRONT OF A SIDE-ADDRESS MICROPHONE.

One of the most important parameters of a microphone to consider when recording is the polar response. Some mics are very directional, others pick up sound equally from all directions, and the polar pattern describes the shape of the pickup area that emanates from the microphone capsule.

Cardioid microphones (with a heart-shaped polar response) are most common, but you'll also encounter omnidirectional, figure-of-eight, hyper-cardioid and super-cardioid varieties. Some microphones even have a switch that lets you choose the polar response. These terms describe the polar behavior, but each mic will have a unique polar pattern, and even an end-address mic will 'hear' sound from behind the capsule, by virtue of the polar pattern. Take a moment to look up the polar pattern of the mics in your collection, so you can make creative decisions with mic placement.

GABRIELLE GRAU

Lyon, France

French solo artist Gabrielle Grau started making music in 2009, after her mother suggested that she play guitar. Finding her inspiration from the streets of Lyon, she started recording herself singing and playing guitar. “I realised that my friends and my parents liked it so I continued and continued.”

Gabrielle has used YouTube to grow her career. “I started a YouTube channel a few years ago because I wanted to get advice from other people, and that’s why I started recording my music.”

“I STARTED A YOUTUBE CHANNEL A FEW YEARS AGO BECAUSE I WANTED TO GET ADVICE FROM OTHER PEOPLE”

SHOCKMOUNTS & POP SHIELDS

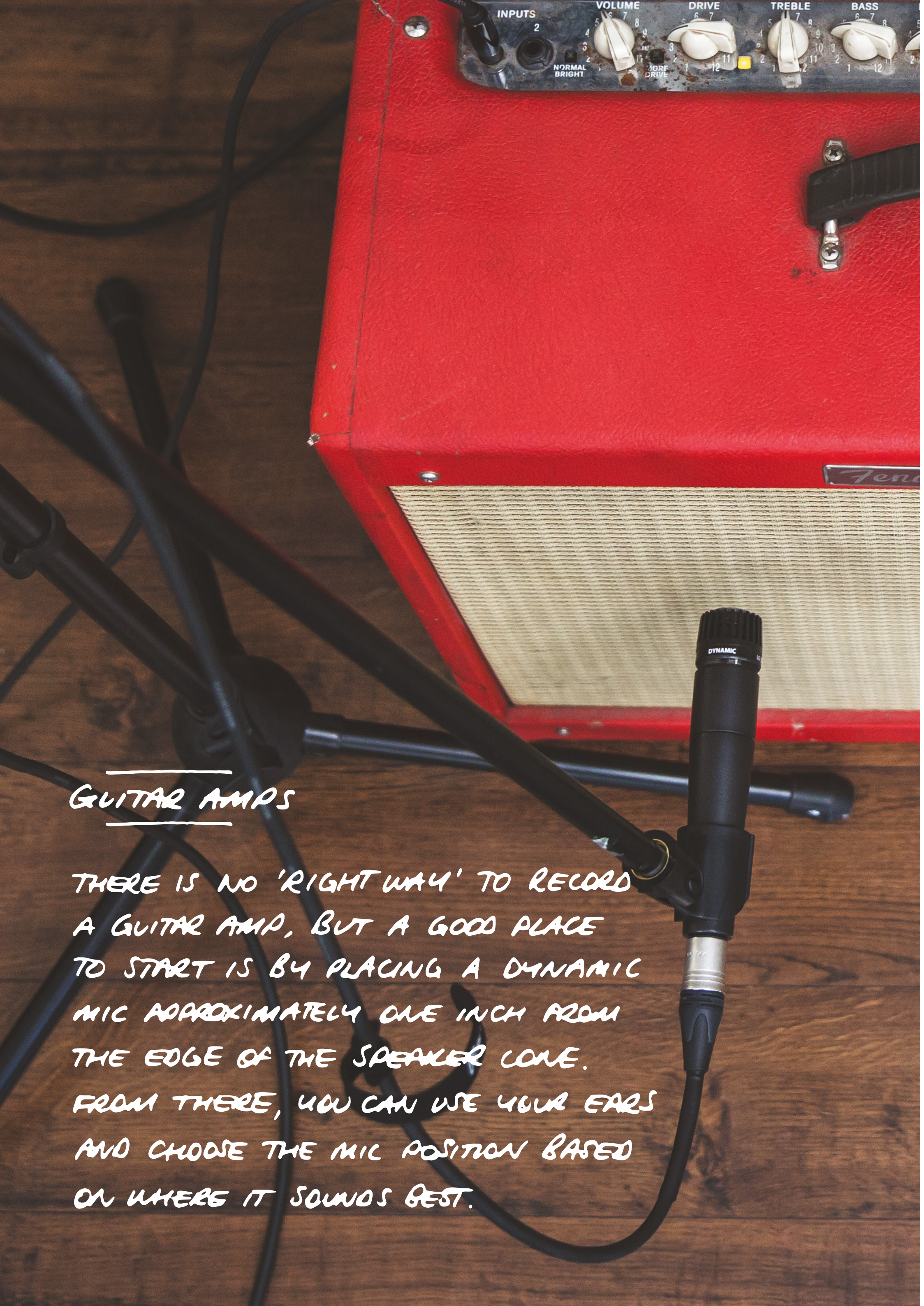
Handling noise and stand-borne vibrations can affect the clarity of a microphone's signal. They're hard to hear and even harder to track down, but oscillations travelling up mic stands from non-solid floors can taint your recordings with a low rumble that clouds your bass ranges. Most condenser mics come with a shock mount, which suspends the mic body in an elastic cradle. It's important to use these, to prevent unwanted rumbles and bumps. If you have hollow floors, you might want to invest in some good shock mounts for your other sensitive mics — especially if there's unexplained low-end activity on your recorded tracks. You can, of course, high-pass filter your mics, if that's a suitable option for your sound source.

Pop shields are another useful tool to prevent unwanted noise making its way into your music. A pop shield acts as a buffer to plosives, which are the pressure waves created when we pronounce 'p' and 'b' sounds. A stray plosive hitting your mic capsule can destroy a vocal take, so a pop shield can be a session-saver.

RECORDING GUITARS

When recording guitar at home, there are lots of choices in terms of approach. Acoustic guitars without pickups will obviously need to be miked up. Acoustic guitars with pickups can be plugged directly into your audio interface — several inputs on Focusrite interfaces will have an 'Inst' feature, which presents a high-impedance input for instruments. But even when pickups are available, many engineers still choose to use mics to record an acoustic, to capture the whole sound of the instrument. Every guitar will sound different, but in general, the sound from the body is more bassy and boomy; the sound from the neck has more high-frequency content, including fingering noises from the fretboard.

In terms of which kind of mic to use, lots of recording engineers will opt for a single small-diaphragm condenser mic positioned between the sound hole and the fretboard. If recording in stereo, pay close attention to phasing issues caused by timing differences at the microphone capsules — an X-Y pair can help here, as the microphone capsules will be at roughly the same point in space.

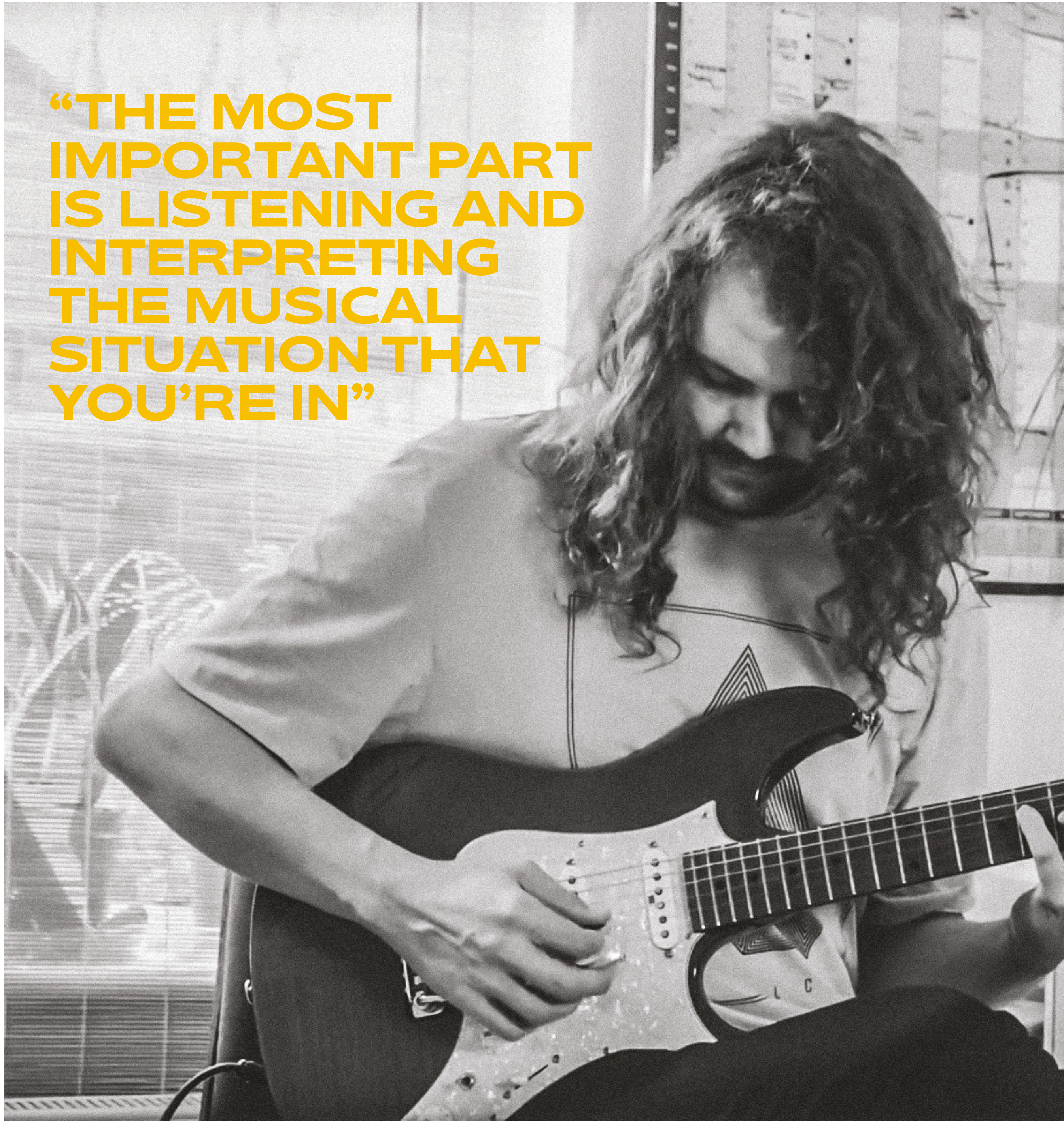


GUITAR AMPS

THERE IS NO 'RIGHT WAY' TO RECORD A GUITAR AMP, BUT A GOOD PLACE TO START IS BY PLACING A DYNAMIC MIC APPROXIMATELY ONE INCH FROM THE EDGE OF THE SPEAKER CONE. FROM THERE, YOU CAN USE YOUR EARS AND CHOOSE THE MIC POSITION BASED ON WHERE IT SOUNDS BEST.

When recording electric guitars, there's the 'old way' and the 'new way'. The old way involves putting a guitar cabinet in a room by itself, cranking up the amp, placing a mic or two in front of it and closing the door tightly. Guitar amps, especially tube ones, do something magical when they're turned up, so many old studios have an isolated amp room where guitar amps can be put through their paces without disturbing the sound in other rooms. In this scenario, the best advice is to use your ears to inform where to place the mic in relation to the amp. Ask the guitarist to play through the track, stick some some closed-back headphones on, and listen to how the sound changes as you move the mic(s) around. This, understandably, might not be an option in your music-making environment, especially if there are neighbours, significant others, parents or flatmates in earshot. That's where the 'new way' comes into play: using amp-modeling software in a computer to emulate the sonic magic that an amp provides.

As with electro-acoustic guitars, you can plug your electric guitar straight into your interface, ensuring the 'Inst' function is selected on the preamp. With the low-latency monitoring afforded by modern interfaces, it's totally possible to use your DAW as a real-time amp modeller.



"THE MOST IMPORTANT PART IS LISTENING AND INTERPRETING THE MUSICAL SITUATION THAT YOU'RE IN"

SAM BELL

England, UK

Sam is a professional guitarist from the south coast of the UK. While he spends a lot of time in recording studios on the 'other side' of the glass, he prefers a stripped down and simple home studio setup. He feeds his guitar into his Boss GT1000 effects processor, then directly into his Focusrite Scarlett Solo, which is connected to a Windows computer running Cubase. "We take it for granted: we can actually record in our bedrooms now! With the Focusrite stuff, you just plug it in and off you go."

Wherever he's making music, his approach remains the same, "The most important part of being a professional guitar player is listening, and interpreting the musical situation that you're in."

Inspiration for Sam comes in equal parts from home and away. "Where we live can inspire certain aspects of your musical personality. I've always loved the south coast of England, and it always fascinates me to look out into the ocean. But we can all relate with finding influences in different cultures. I think it's a very healthy thing to do. I'd like to think that going to these places gives me some headspace to get some new ideas."

MIKING DRUMS

Recording drums is one of the most challenging aspects of music making. There are an infinite number of approaches, but the first things to consider are the tuning of the kit and the space you're recording in. A poorly tuned kit in a dull-sounding room will never sound good. Don't be afraid to dedicate a lot of time to finessing the drum setup. Experiment with micro adjustments to mic positioning; the sound of a snare drum will sound drastically different over the space of a couple of inches, for example.

OVERHEADS

CONDENSER MICS PROVIDE AN ACCURATE AND CLEAR REPRESENTATION OF THE SPACE THEY'RE PLACED IN. FOR THIS REASON, WE USUALLY USE CONDENSERS AS OVERHEAD MICS.

SNARE DRUMS

A DYNAMIC MIC POSITIONED 2" FROM THE RIM OF THE BATTER HEAD WILL USUALLY GIVE YOU GOOD RESULTS, BUT KEEP YOUR EARS OPEN FOR PINGY-NESS AND A LACK OF DEPTH. DON'T BE AFRAID TO MIC THE RESONANT HEAD OF THE SNARE – JUST WATCH OUT FOR PHASE ISSUES

When it comes to miking, you might choose to close-mic everything (where you place a dedicated microphone on every single drum), or cover the overall sound with some well-placed overheads. Often a mixture of both will give good results. It's conventional to mic the batter head of a drum (the side that gets hit by the drumstick).

When close-miking, bear in mind that the rim of the drum can sound very pingy; positioning your mic an inch or two away from the rim, towards the centre of the drum, will help to get a chunkier sound. But don't overlook the resonant head (the underside of a drum), which can provide a different tone, and thicken the overall drum sound. Some people mic the underside of the snare, to pick up the snap and sizzle of the snare wires. But watch out for phase issues here — you might want to flip the polarity on your mic preamp.

THE KICK DRUM

We could write a whole book on miking kick drums. Some engineers like to remove the resonant head, stuff the drum with a couple of blankets and mic the kick from a few feet away. This results in a solid thud that's cool for R&B and funk. Others like to leave the res-head on and tune it down to be super flappy. This is popular for jazz and styles where a more natural drum sound is desired. Others use ported drum heads, some open the drum and don't use any dampening at all. It's all totally subjective and all we can advise is to experiment, use your ears and only start recording when you're really happy with the drum sound.

Final tip: when you've used all your input channels on a drum setup and you still aren't happy, Google 'Glyn Johns' three-mic drum technique'.

KICK DRUMS

THERE ARE LOTS OF WAYS TO GET A GREAT KICK DRUM SOUND. EXPERIMENT WITH DAMPENING THE INSIDE OF THE DRUM WITH TOWELS. AS EVER, LISTEN TO THE SIGNAL FROM THE MIC TO MAKE YOUR CREATIVE DECISIONS.

"I JUST WANT TO CREATE, AND I DON'T THINK I'LL EVER STOP CREATING"

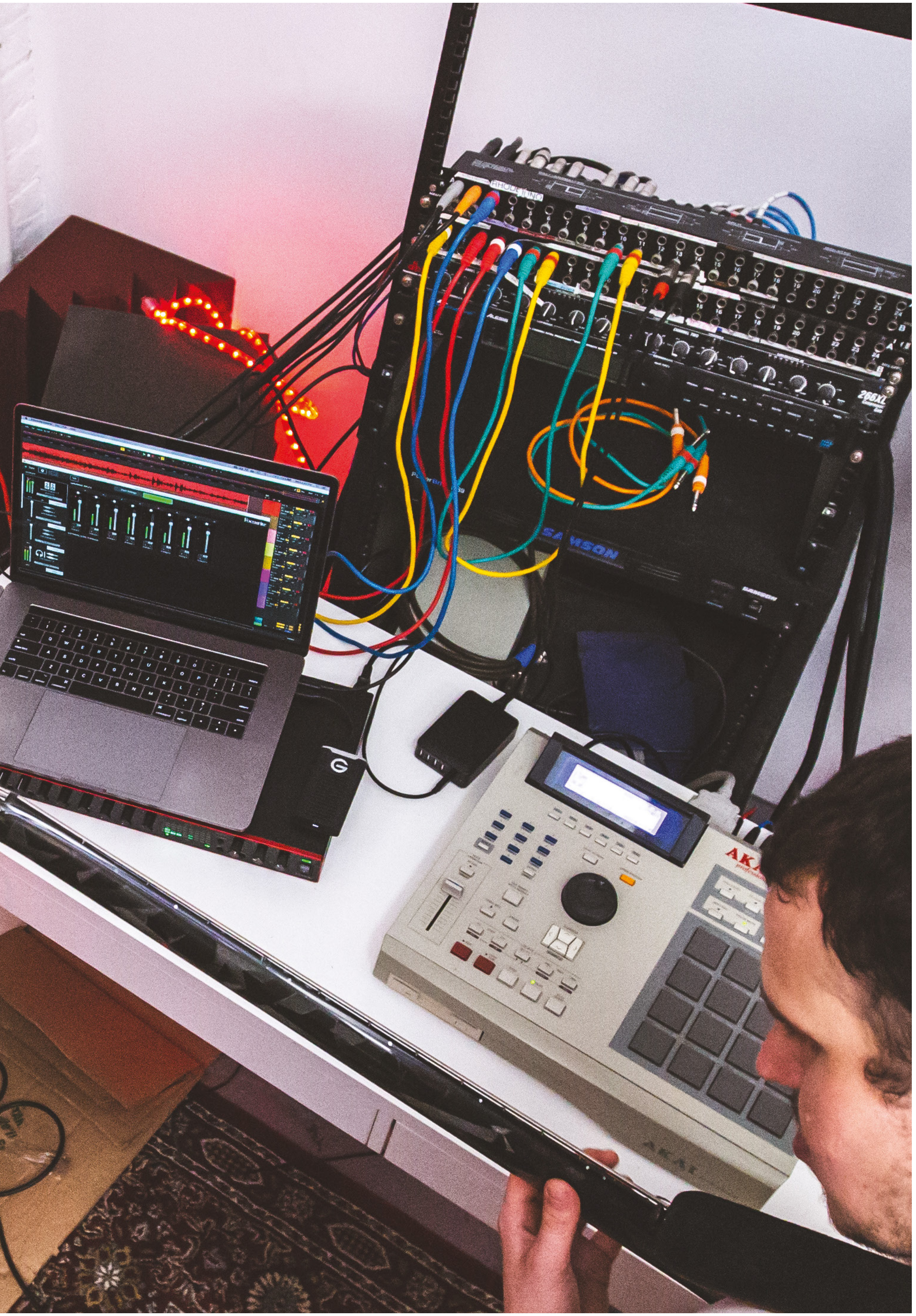
SLOW CLAP

Buckinghamshire, UK

Danny Nugent (aka Slow Clap) is a synth lover and designer, with multiple music projects and creative outlets. "My initial aspiration for my music," says Danny, "was to be able to play live and be in a situation where people wanted to come see me, even if it was 10 or 20 people, but they were there to see me, which luckily I've achieved."

His Scarlett interface gives him lots of flexibility for connecting various instruments such as synths and guitars, and for connecting to MIDI equipment and outboard effects processors using the multiple outputs. "I can bring in different outboard gear into my setup, and it's great that it's got MIDI on it because I can send MIDI from Ableton Live."

Danny has one simple objective in music making: "I just want to create, and I don't think I'll ever stop creating."



MIXERS & PATCHBAYS

For those who prefer to work with a hardware mixer, an audio interface with lots of inputs and outputs can be used to integrate the mixer into the DAW workflow. The benefits of this approach include being able to route audio to things like effects sends and custom headphone mixes, having per-channel EQ, and enjoying one-to-one control over every parameter. The most convenient method to connect a mixer to an interface is to send direct outputs from the mixer to the audio interface line inputs, and send DAW returns from interface outputs to empty channels on the mixer. Of course, you need to watch out for feedback loops, keep on top of your DAW monitor source, and do all you can to keep latency to a minimum.

A patchbay can also be a great addition to a studio, especially where you have your interface in a rack, you work with hard-wired inputs, and only occasionally need to patch, say, a guest's synth into the system. The benefit of a patchbay in this scenario is that you don't need to fumble about in the back of the rack to unplug your cables; just plug into the patchbay to make the connection. A patchbay can be wired so that a 'normal' configuration is established when nothing is patched at the front.

24-BIT RECORDING

Compared to 16-bit recording (the CD-standard), 24-bit recording provides far greater audio resolution and a much more forgiving recording environment. It does consume more data (1.5x more than a 16-bit file), but with terabytes available for cheap these days, the increase will hardly move the needle in terms of storage space. Those additional 8 bits allow for your digital audio to be expressed more accurately: in the 16-bit domain, the amplitude of a single sample can be represented by 65,363 possible values. With 24 bits, there are more than 16 million values, which means more precise audio, greater clarity and better-sounding results! There really is no reason not to record in 24-bit.

BACKING UP

The process of backing up is not unique to music making, but it is a crucial part of the process — to ensure your creations are not lost when a hard drive fails or you leave it in your backpack on the train. Data should be backed up in three places, and the drives should be in different physical locations to avoid all backups being destroyed in a catastrophic event such as a flood or fire. Cloud storage is one option for backing up, as are automatic backup systems such as Apple's Time Machine. Allocating some time every week to backup your files is time well spent.

BACKUPS

KEEP SEVERAL BACKUPS IN DIFFERENT LOCATIONS. YOU NEVER KNOW WHEN YOUR BACKUP WILL BE A LIFE-SAVER!

“HAVING SOMETHING HANDS ON THAT I CAN TOUCH AND FEEL HAS HELPED MY PROGRESSION AS A PRODUCER”

UNGLUED

England, UK

Unglued is a drum and bass producer signed to Hospital Records, who uses a lot of vintage hardware in his music-making process. “I’m investing in a lot of older equipment, and that helps me get that old sound. [Old gear] has a lot more character, and it has an authenticity to it, its own energy.”

But his production setup has not always been conducive to using analogue outboard, as he was restricted to just a headphones port and in-the-box sounds. “Before I got my Scarlett interface, I couldn’t send any audio in or out of my computer.” Now, his vintage tape machine and spring reverb get as much use — if not more — than plugins.

“Having something hands on that I can touch and feel has helped my progression as a producer. That tactile sensation really helps with the creative process.”

TIPS: BUILDING A MODEST FOLLOWING AND RELEASING MUSIC INDEPENDENTLY

Will Evans from **Tape Club Records** is no stranger to the independent music industry. Tape Club specialises in "The Best of British Home Recording" and champions undiscovered music makers. Will has some tips to releasing your own music and building a following.

"Don't be shy. The internet has proven that there is an audience for everyone. You just have to put the work in to find it. For some it will be easy, you might share a private Soundcloud link with someone and they share it with a friend at a label and, hey presto, you're signed. For most, however, it's a grind. You'll have to work to be heard."

"Think of it like exercise. Commit to a daily routine that has you working on sharing your music. It could be sending a WhatsApp to your family, hand-writing letters and sending them to radio presenters, or sending ten Instagram DMs a day to strangers. Some will stick and you'll learn a lot about yourself and your art along the way."

"Help your chances by connecting yourself to others in the music making community: other artists, DJs, engineers, promoters. Don't hassle them, just make friends. Over time this can provide you with an organic network to share your music with."

"It might happen overnight, but it probably won't, so be prepared to buckle up and enjoy the ride. It'll be a long one, but if you've got your priorities right, it'll be a fun journey irrespective of the outcome."

CREDITS

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Thanks to: Cesar Toribio, Lauren Martinez, Scott Kapelman & Teddy Mars.

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