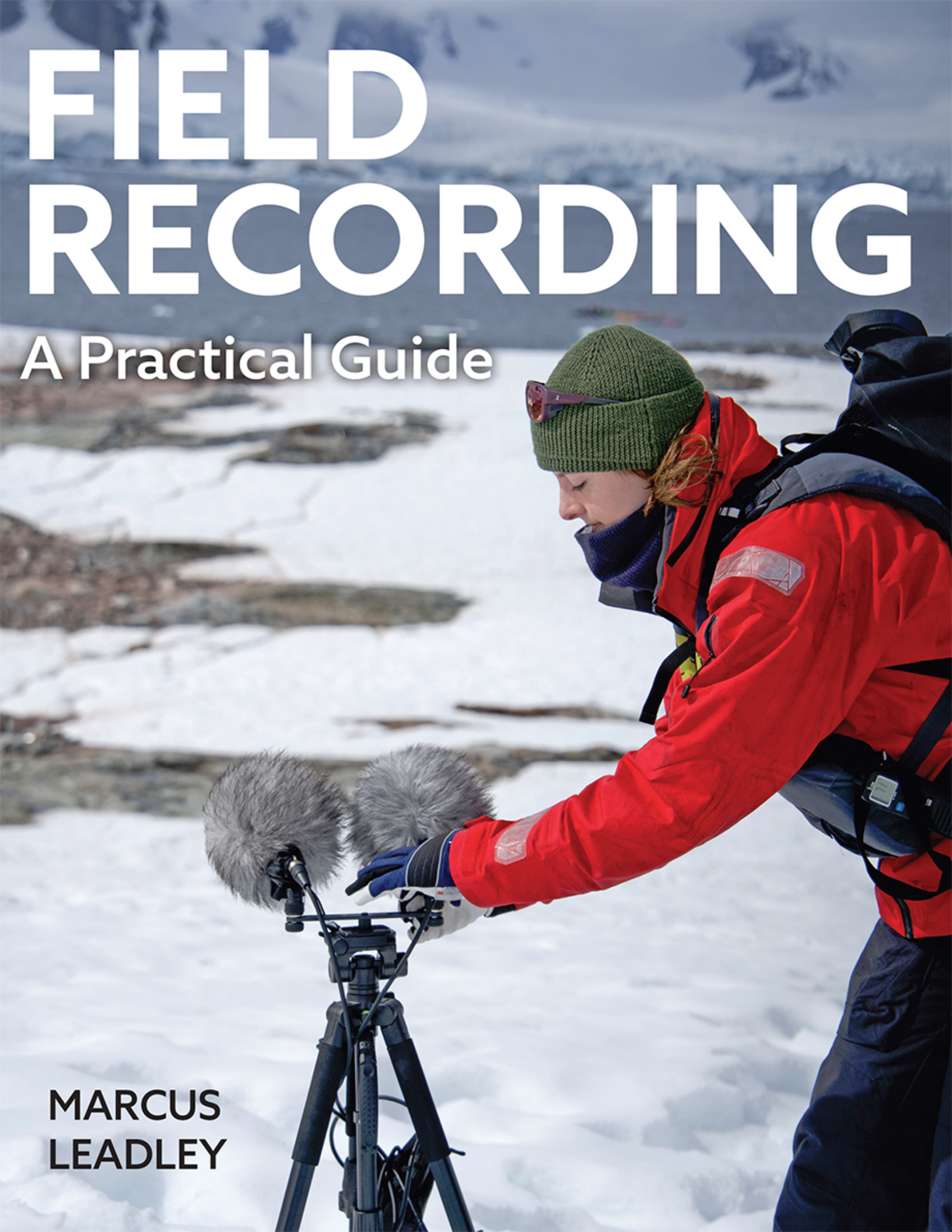
FIELD RECORDING

A Practical Guide



MARCUS LEADLEY

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INTRODUCTION

This book is designed as a practical guide to field recording for people interested in documenting all forms of environmental sound and recording wildlife. It's also for musicians and artists who want to use field recording and found sound in creative practice and researchers, podcasters and other practitioners for whom field recording is becoming an increasingly important part of media production and documentation. I address all the key basics before focusing in on advanced topics as we progress through each chapter – so this book is for you if you're a complete beginner or have some experience of recording with your phone or a hand-held digital recorder. And there's plenty of content for those who've already honed more advanced skills using professional equipment. Note, however, that this book isn't intended as a training manual for any particular profession, so I won't be exploring specific trade practices. My primary objective in writing it has been to help you develop a depth of knowledge that will enable you to make creative choices about what, how and when to record.



While the main focus is on recording airborne sound, I'll also be introducing a range of tools and techniques for recording in environments you may have considered inaccessible – such as capturing underwater sound. I've tried to keep to basic, plain English explanations rather than rely on equations; sound, transmission, diffraction, reflection and attenuation are complex phenomena and there are plenty of texts and resources for those who wish to follow the science. Here, my aim has been to use the science where

necessary to explain the reasons why some things happen, and others don’t.

So what is a field recording? I always like to propose the broadest possible definition: any recording that takes place outside the specifically prepared environment of a recording studio. Field recordings, almost unintentionally, capture the acoustical traces of landscapes, locations and populations; the potential intertwining of topography, meteorological phenomena, architectural acoustics, fauna, flora and mechanical processes. That’s a very different proposition to a studio recording where we usually aim to isolate sound and music from the outside world in the pursuit of control and fidelity. Yes, we might want to capture the reverberation of an acoustically engineered room to enhance the recording, but nothing else. In the field on the other hand, we are dealing with the sound of the world exactly as it is. That doesn’t mean we don’t make creative and technical choices to achieve the best results in terms of clarity, focus and content capture but we proceed from the situation as we find it. This is one of the things that makes field recording so exciting. Whether it’s the soundscape of an urban night, a tropical rainforest, a wire fence resonating in the wind or insects moving under the bark of a tree, an understanding of the right tools and the appropriate technique is the key to achieving great results.

The act of listening lies at the very core of field recording practice and the two enjoy a wonderfully reciprocal relationship. Listening informs field recording choices and decisions and the act of recording expands our capacity to focus and really listen to the world around us. This enables us to comprehend depth and nuance in new and exciting ways. Throughout this book I champion the primacy of listening and explore it through the experiences of numerous practitioners and artists.

Chapter 1 kicks off with a deep dive into the world of field recorders and best practice. There’s an in-depth look at the range of products on the market, and this is something you’ll find throughout this book; the world of kit is extremely confusing, especially when you first start looking at options. I spend a huge amount of time

advising students, especially keen adult learners, on the equipment they need to achieve their goals, and this always comes with a budgetary consideration. So you won't find me always championing the 'reassuringly expensive' gear; I always consider options at every price point. While what's presented in this book isn't intended to be an exhaustive overview of the equipment on the market, I have tried to cover a wide range of manufacturers and products. Where I include price information this is purely as a generalised budgetary guide as prices of individual items will fluctuate. It's also worth noting that while products will be discontinued and replaced with different or updated units, none of the equipment featured in this book is about to become obsolete. Microphone technology has essentially been stable for decades and any file-based digital recorder will continue to give good service all the time it remains in good working order. And today's models are simply tomorrow's second-hand bargains!

Chapter 2 focuses on the nitty-gritty of capturing airborne sound with phones, hand-held all-in-one recording devices and specialist microphones. So I look at the benefits and uses of the many different types of microphone and how to set them up to get the best results. Dealing with wind noise gets a special focus and I explore ways of working in more extreme environments.

Chapter 3 groups together a range of equipment and techniques for exploring what I call 'hidden' sounds: sounds you can only hear with specialist equipment. I'll be covering underwater sound (hydrophones), sound outside the range of human hearing (ultrasound and infrasound detectors), sound transmitted through surfaces, objects and structures – including trees and plants (contact microphones) and the rich world of electromagnetism that bathes every aspect of contemporary experience with its extraordinary tones and sonic textures (coil microphones).

Chapter 4 explores multichannel sound recording. Surround sound is currently going through a phase of rapid development with Dolby Atmos-enabled technologies now in the home and Ambisonic diffusion becoming a regular feature of sound design and gallery experience. Field recordists and content creators have access to a

range of exciting new tools for making and delivering work for these environments.

Chapter 5 focuses on the advance planning that makes recording field trips an enjoyable and productive experience. There's also practical information on both equipment and data management in the field and a list of questions you might like to consider when planning a trip. Plus, there are plenty of professional tips and tricks that will help you save time on the ground and make the recording experience as enjoyable as possible.

Chapter 6 is all about what you do with your recordings once you get back to base. This section about studio practice covers everything from file management and the basics of working with digital audio workstation (DAW) software to the external peripherals (sound monitors, interfaces, headphones and so on) that make working with environmental sound practical and enjoyable. Of all the chapters in this book, this one could easily expand into a separate book! As a result, I stick to basic needs and principles given the space available. So, for example, you'll find plenty of information about software options and general capabilities, but I don't go into the operational details of specific software. There's an overview of the types of plug-ins that are useful to field recordists and some pointers in the direction of the more widely used studio tools.

Chapter 7 is something of an added bonus for a book that's dedicated to practical advice; the history of field recording is a fascinating tale about technological breakthroughs, famous early adopters, unlikely expeditions and the developments that have made field recording equipment and know-how accessible to everyone.

Writing this book has given me a fantastic opportunity to engage with field recording as a multifaceted and focused enquiry. It's a discipline I've been passionate about ever since I was a young adult managing a hospital radio station in the early 1980s and it's led me on a creative journey through various aspects of professional practice and into academia as an artist, researcher, technician and facilitator. Hopefully my enthusiasm for field recording comes across loud and clear and I hope this book will encourage you to embrace

the richness of the acoustic environment and help you explore, record and enjoy – wherever that journey takes you.

Before we start reading about field recording, we shouldn't forget the importance of simply listening. So, to that end, let me also point you in the direction of my website (www.marcusleadley.com) where you'll find an ever-expanding range of recordings, soundscape compositions and more. If you're interested in my online field recording classes or practical field trip sessions (where I provide a wide range of equipment to try out), details can also be found at www.marcusleadley.com

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1
DIGITAL FIELD RECORDER BASICS

It’s only been a century and a half since Thomas Edison first captured the sound of the human voice. By the turn of the twentieth century, his wax-cylinder phonographs were already becoming standard kit for scientific and ethnographic expeditions. After this, field recording migrated to disk-based systems and the pursuit of sound quality ushered in an era when heavy and hugely expensive equipment (often built into cars and vans) became the property of record companies, national broadcasters and a handful of researchers with academic support. Next came tape-based solutions, increasingly portable high-quality machines and new, lower-cost devices aimed at the mass market. These opened the field to an entirely new generation of professionals and enthusiast ‘sound hunters’. Then came the moving media digital formats such as DAT and minidisk. Digital file-based recorders started to appear in the mid-1990s, and by 2010 had become ubiquitous.



Recording everyday sound on a budget: Roland R-07.

Today, there are many different types of lightweight digital file recorders to suit different budgets and needs. There's a good range of small hand-held devices offering all-in-one recording solutions as well as stand-alone recorders for inclusion in more advanced recording kits. Many of the low-cost devices feature built-in microphones and have limited connectivity for additional mics. Intermediate-price solutions continue to offer the convenience of (higher-quality) built-in microphones – but also allow users to

connect a range of external microphones/devices. Some will double as a computer interface. Higher-quality/cost field recorders rarely feature built-in microphones; the assumption here is that the user will be making choices about the microphones they want to use and will prefer to purchase a recorder where the cost reflects a focus on delivering the best build quality, circuitry and features.



Zoom H1essential.



Tascam Portacapture X8.



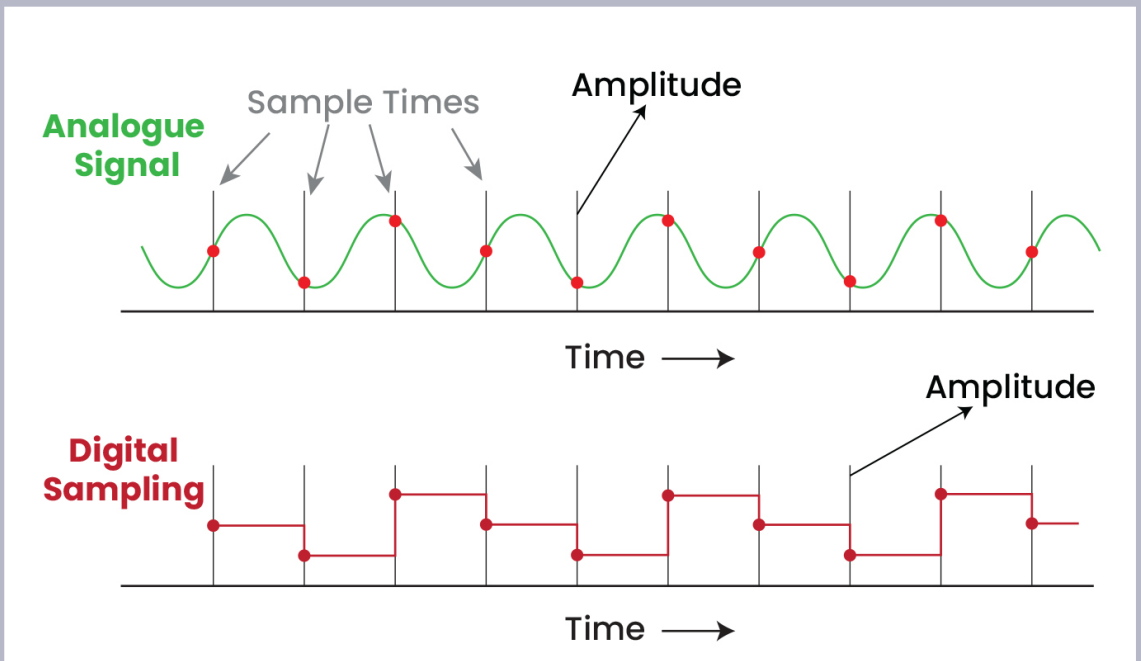
Sound Devices MixPre-3 Mk II.

Zoom H1essential, Tascam Portacapture X8 and Sound Devices MixPre-3 Mk II: examples of contemporary digital field recorders to suit all needs and budgets.

While designs, model numbers and feature sets will change over time, and new technologies emerge, the basics of digital recording aren't about to change. This is good news; while we may become aware of the limitation of a piece of equipment and choose to upgrade, the current options at all price points will be capable of delivering many years of good service. In this chapter I'll look at recorder basics, best practice and explore the advantages and disadvantages of each category of field recorder. I'll also look at some of the more popular brands and models.

Factfile 1.1: Analogue vs Digital Recording

Old-fashioned analogue tape captures sound as a continuous magnetic trace. Hence a high-quality analogue recording can be very accurate. Unfortunately, over time, tape degrades and magnetic traces can be compromised. And analogue copies lose fidelity at each stage of removal for the original recording. So, making a copy of a copy will amplify any system noise and high-frequency content will be lost. Digital signals, on the other hand, are comprised of discrete elements: samples. Each sample reflects a set of values at a specific moment in time. Once numerically encoded, these remain constant and can be copied an infinite number of times without degrading. This doesn't mean that every digital audio file will deliver maximum fidelity; there is always some trade-off between sound quality and file size. This is why settings can be adjusted and there are different digital file formats for different applications.



Analogue VS digital signals. Note how lower sample rates introduce inaccuracies.

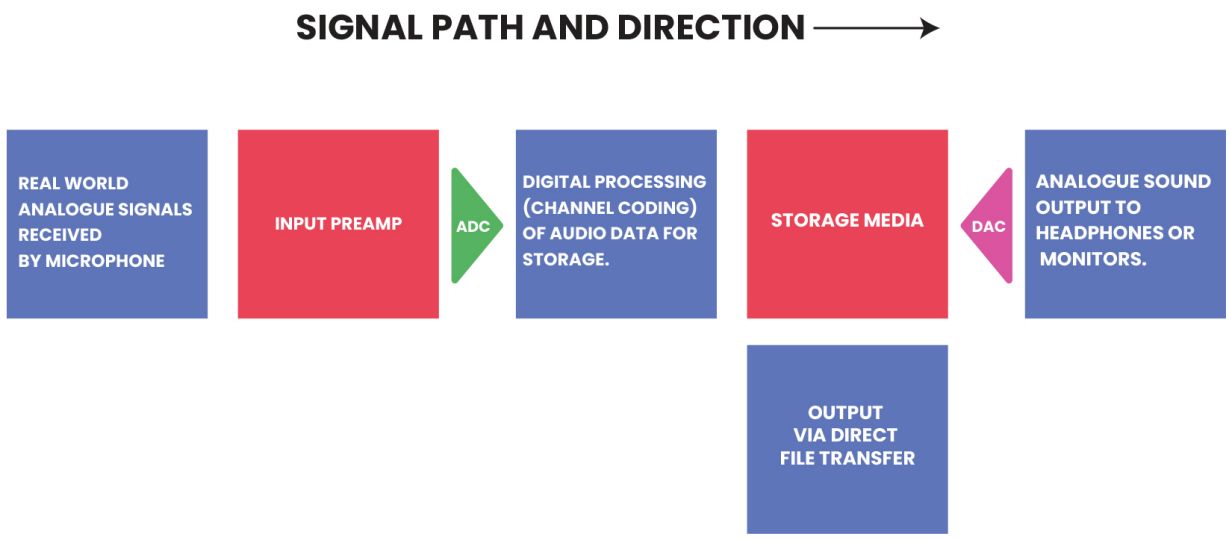


Uher 4004 Report Monitor (analogue tape) and Tascam DR40. Analogue vs digital; a big difference in size and portability.

FIELD RECORDER BASICS

Signal Path

All digital audio recorders seek to capture sound in realistic detail. To do this, various stages – from the arrival of sound at a microphone to digital encoding and data storage – need to be carefully managed. As the path is dependent on physical processes, circuitry, software, materials and manufacturing methods, decisions will have been made in terms of cost versus quality all the way along the line. For the most part the old adage ‘you get what you pay for’ applies here – but that doesn’t mean that lower-cost units can’t deliver exceptional results.



Block diagram of recording signal chain.

Microphone/s

At the most basic level a microphone is a transducer – a device that converts sound into an electrical signal. We’ll be looking at

microphones in much greater detail in Chapter 2, but for the time being this simplified definition will serve. Sound is a sequence of pressure waves that propagate through compressible gasses such as air, liquids or solids. When the energy of the wave arrives at the microphone it causes a diaphragm to vibrate – and it's this vibration that's converted to electrical energy. Microphones need to work efficiently with the next step of the chain, the recorder's microphone preamp, to maximise power transfer and preserve frequency content. This is achieved by 'matching' the low output impedance of the microphone with the input impedance of the preamp (*see* Factfile 3.2 for more information). With built-in microphones, this relationship is designed in. For obvious reasons, the specifications of most of the external microphones we are likely to use for field recording and the full range of field recorders are designed to be complementary. High impedance piezo-based contact mics and hydrophones are an exception, and we will look at these in Chapter 3.



XYH-6: one of a range of interchangeable mic capsules for Zoom recorders.

Preamp

The preamp takes the low-level output from a microphone and amplifies it. Much of the cost of the more expensive recorders is associated with preamp design and component quality, so expect greater efficiencies, which are reflected in sound quality. High-quality microphone preamps will be high-gain units, which is ideal when recording quiet sounds. They will have a low noise floor (so less of the hiss you hear in the background of the recording that's generated by the recorder itself), deliver a tonally accurate character over a wide dynamic range and, if properly set, plenty of headroom. Headroom is the 'distance' between the recorded signal and the onset of digital clipping at 0dBFS – often referred to as the distortion ceiling. Low-cost preamps can't compete in terms of gain and will have more self-noise. However, your budget is going to be the

deciding factor here – or you may simply not want to invest a lot of money at this stage if you're new to field recording. The good news is that the well-designed preamps of even the entry-level hand-held devices perform remarkably well. Note that much of the above relates to the preamps of the contemporary recorders designed along 'traditional' lines which are still in the majority; we'll be discussing the specifics of the alternative 32-bit floating point technology in a separate section later in this chapter.

Analogue to Digital Conversion (ADC)

Analogue sound is continuous, physical vibration and it needs to go through a conversion process in order to be stored as digital information. It needs to be sampled – listened to or described – at regular intervals. Contemporary devices use Pulse-Code Modulation (PCM) to do this: either integer PCM (samples are represented by whole numbers) or floating point PCM (decimal points and exponents are used to allow for the handling of values with fractional parts and a wide range of magnitudes). With integer PCM we need to set the sample rate and bit depth (*see* Factfile 1.2 for an in-depth explanation). Higher sample and bit rates deliver more accurate, higher-quality recordings. However, this means larger files. This is less of an issue now than it was in the early days of digital recording due the relatively low cost of SD memory cards and other storage media. With floating point PCM we still need to set the sample rate, however 32-bit is the default standard. In both cases, once the analogue sound is converted to audio data it is digitally processed for storage.

File Format

Digital recorders based on integer PCM processes allow us to select different file formats for recording and the most common are Waveform Audio (WAV) and MPEG-1 Audio Layer 3 (MP3). WAV is an uncompressed, lossless format and MP3 is a compressed, lossy

format. In this context, compression refers to a process that's designed to generate smaller file sizes. The simple take-home message here is always use an uncompressed, lossless format for field recordings – unless for some reason you only have very limited data storage capacity and/or capturing content is more important than fidelity. For example, recording a long lecture. MP3 can still sound remarkably good and the small file size is necessary for many audio-streaming or online distribution applications. However, the information content is significantly reduced (*see* Factfile 1.2). If you're planning to use plug-ins or any other form of processing in postproduction, make sure you're recording high-quality, lossless audio. And you can always convert from WAV to MP3 later, as required. You can also do the conversion the other way around, but you won't gain any audio quality. For both WAV and MP3 files you can specify a higher or lower recording quality. With WAV this means selecting the appropriate sample and bit depth; for MP3 you select kilobytes per second (KBPS): the bit rate. In both cases, higher numbers always equate to higher-quality audio – but this means larger file sizes.

It's my feeling that 48kHz/24-bit is still an acceptable standard for documentary field recording, collecting material to use in soundscape composition and in the development of materials for music and home-brew special effects. And this won't break the digital storage bank! If you're planning some extreme processing, especially downward pitch shifting, you may want to experiment with 96kHz/24-bit source files to ensure that you've recorded high-frequency content to bring down into the audible range. Bear in mind however that taking advantage of higher-fidelity audio files means more data usage at every stage of your process: bigger sessions on your computer and more drive space for backups. Older computers may also struggle with the extra CPU demand.

Factfile 1.2: Sample Rate and Bit Depth – The Deeper Dive

The terms sample rate, bit depth and WAV file began to enter popular consciousness in the early 1980s when CDs were introduced; 44.1kHz/16-bit WAV was the file format for the medium. WAV files can have both higher and lower sample/bit rate than this and these factors determine audio fidelity. The sample rate defines how often the sound is described, measured or recorded (in other words sampled). For the old 44.1kHz standard this was 44,100 times per second. The higher the sample rate, the closer you'll get to the 'smoothness' of the original analogue sound. The sample rate also dictates how wide a frequency range you can record. The best-case scenario for human hearing (high-frequency hearing loss begins naturally at around the age of 18) is 20Hz–20kHz. Most 40-year-olds cannot hear above 15kHz. To capture and then reconstruct a specified bandwidth, the sample rate needs to be at least double the highest frequency. Hence, we can see why 44.1kHz became established as the first standard, balancing adequate fidelity and minimum data requirement.

Do we need to record sound outside the range of human hearing? We'll certainly want to record ultrasound if we are recording dolphins, bats and some species of whale that use it for echolocation – and in any other situation where we are planning to shift recordings down into the range of human hearing – otherwise we will have no content to convert. And we might want to do this more generally for material we plan to use in composition or for making sound effects. At a more general level, ultrasound has a role to play in the subtle psychoacoustic effects that contribute to how sound is perceived, not simply heard.

The bit depth establishes how great a range of volume increments can be described for the sound being sampled. It

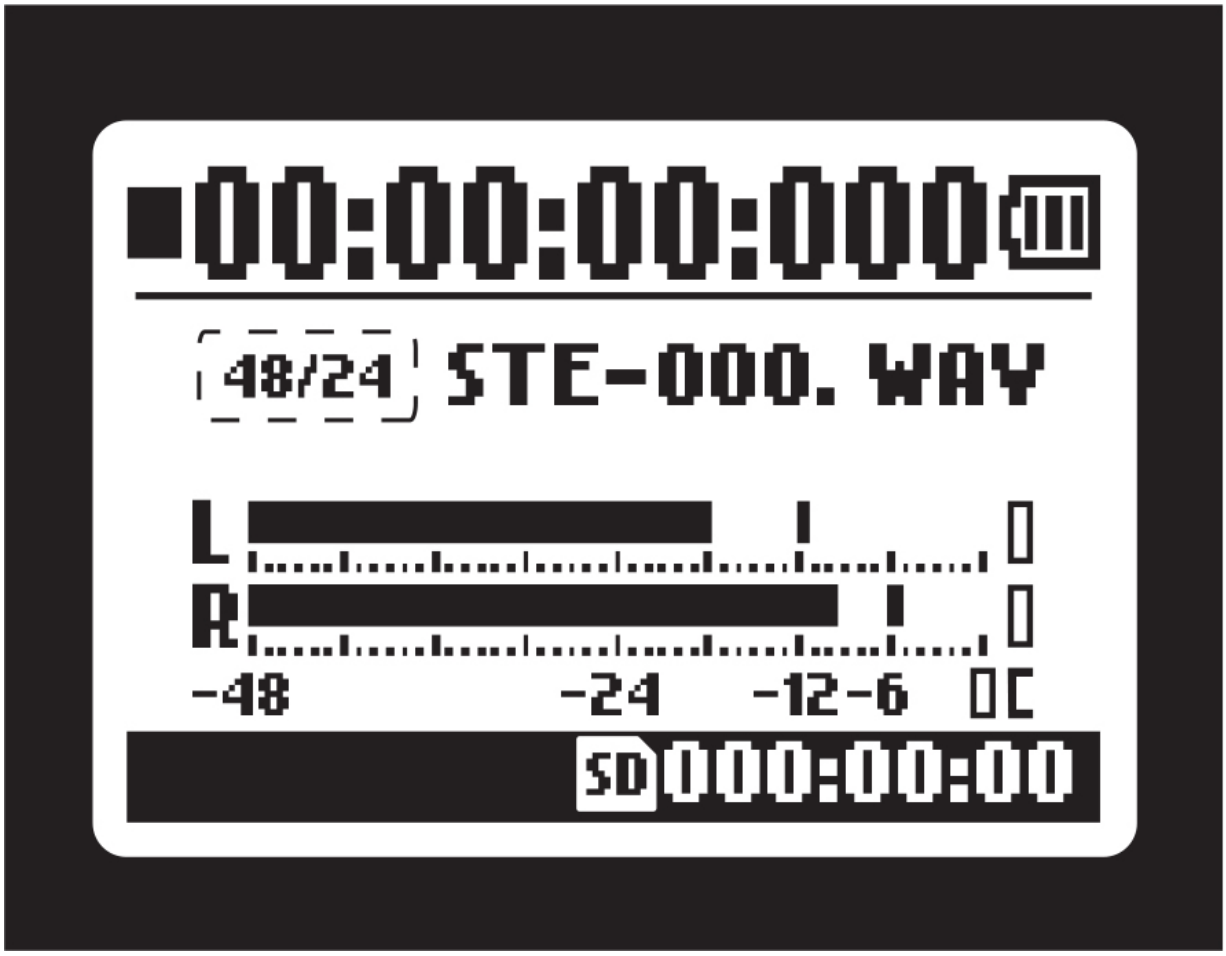
basically describes how many digits are used to store each sample of the analogue signal. Higher bit depths give a more accurate representation. Again, the old CD standard of 16-bit audio offered 65,536 volume increment options (2 to the 16th power). When CD was the dominant commercial carrier there were plenty of field recordists who could see no point in using higher-fidelity recording options (even though DAT audio tape could record at 48kHz) because ‘hey, it’s going to end up on a CD anyway’. Post-CD, we find ourselves in a world where commercial listening formats such as MP3 and M4A offer a significantly reduced level of audio fidelity. Thankfully, the attitude of most professionals now reflects the underlying belief that the best way for sound to survive the transition to MP3 is to start with the highest resolution file possible. Most field recording professionals use 48kHz/24-bit which is the current industry standard (so sampling 48,000 times per second and offering 16,777,216 [2 to the 24th power] volume increments). With the general availability and falling cost of high-capacity storage media we can now also consider 96kHz/24-bit recording or even the 192kHz/24-bit option for ultra-hi fidelity work.

Bit depth also determines the dynamic range – this is the distance between the quietest and loudest signal that can be recorded. If the loudest signal exceeds this threshold, the result is audible clipping. If it’s below, there is headroom. The dynamic range for 16-bit is approximately 96dB, for 24-bit it’s 144dB. This all needs to be put in context; in the real world, sound rarely exceeds 120dB – think a massive thunder clap directly overhead or someone firing a shotgun in a nearby field. So surely 24-bit recording is all you’re ever likely to need? The problem is those pesky quiet sounds; in order to capture them you have to add gain from the mic preamp, and this squeezes the dynamic range. So, when you’re recording those quiet insect sounds in the hedgerow and a motorbike goes past, the noise exceeds the dynamic range and the signal clips. This

brings us to the merits of 32-bit float recording – more about this shortly.

Signal and Level Management

The meter on an integer-based PCM digital field recording device is calibrated to show a scale up to 0dBFS (decibel full scale), the highest signal level that can be achieved before the onset of clipping. People often incorrectly refer to this marker as 0dB, which is essentially meaningless, because all uses of the decibel require some form of qualifier. For example, 0dB SPL (sound pressure level) is used to indicate the quietest sound that is perceivable by the human ear. When digital recordings exceed 0dBFS (unless using 32-bit float technology) the recording will start to clip and distort. This may be heard as audible clicks at the point of peak signal level. In more extreme cases, for example when a gust of wind hits an unprotected microphone, the audio content will be entirely obliterated. Isolated clicks can often be dealt with in postproduction but avoiding digital distortion in the first instance should be one of the field recordist’s primary aims.



Digital meter showing ideal recording level between -12 and -6 DBFS

So what is best practice when it comes to setting levels? Ideally, we want our meter to read as close to 0dBFS as possible but not exceed this, even momentarily. At the same time, we want a good, healthy signal level; low-level signals generate their own problems – some level of system noise is an inherent characteristic of all recording devices based on integer PCM technology. While the digital medium itself doesn't contribute (so none of the tape hiss we had in the analogue era), microphones and preamp combinations deliver a degree of self-noise. Indeed, this is one of the major differences between budget and professional equipment; minimising a unit's noise floor requires a no-compromise approach using the highest-quality components. If the incoming signal is recorded at a low level it will be closer to the recorder's noise floor and this will

become apparent in postproduction when we attempt to lift the gain in order to deliver a good listening level.

What we need to aim for in the field is the best possible signal-to-noise ratio. This is achieved by keeping the meter level between -12 and -6dBFS whenever possible. This is relatively easy to achieve in relation to recording music, urban locations and close mic'ed details. However, rural and wilderness environments can be highly problematic. It's here that budget equipment really struggles to deliver detail in low-level signals, due to microphone sensitivity/quality and recorder self-noise. By comparison, high-end solutions offer a lower noise floor and higher gain mic preamps. This doesn't mean that budget solutions cannot be used in these environments, simply that one cannot expect the same level of pristine sound quality. However, 32-bit technology is designed to function very well with low-level signals and the release of budget and intermediate-priced solutions is making it accessible for more enthusiasts.

On the subject of system noise, there is an issue here concerning perception that relates to monitoring in both the field and the postproduction/listening environment. In both situations there is a tendency to listen to environmental recordings too loud: in the field to mask the actual environmental sound coming from beyond our headphones and in post when, without visual stimulus, it's tempting to turn up the volume beyond what was present in the actual environment to create impact. When we do this, we also turn up any recorder noise so hiss can seem more apparent than it actually is. Here it can be useful to use a sound pressure level meter to take ambient readings in the field and establish the same SPL in the listening environment.

32-Bit Float Recording

Professional recorders have offered 32-bit floating point recording as an option for some years and it's now becoming firmly established in the entry level and intermediate product arena. Zoom, for example,

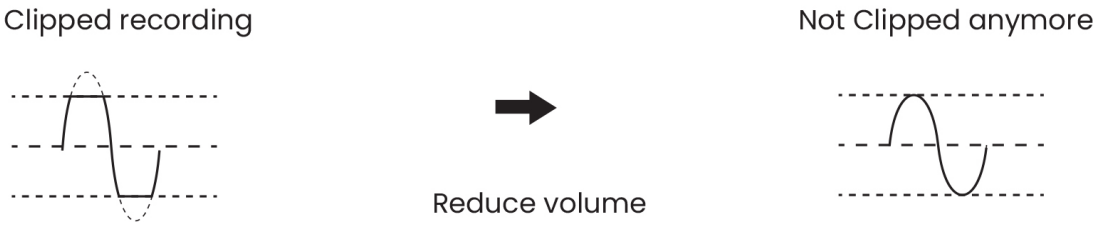
is betting heavily on its 32-bit technology and has replaced its older H1, H4n and H6 models with new 32-bit only units. A new H2essential has also been released and there's a new 'pro-grade' handy recorder, the H5 Studio. Tascam has introduced the FR-AV2 with a similar footprint to the Zoom F3 but offering 24-bit recording as well as 32-bit and enhanced timecode functionality for syncing with cameras. Deity is offering the PR-2, which can also generate or sync to timecode and supports mono and stereo mics on 3.5mm plugs.

Fundamentally different to the fixed-point recording we have been talking about so far, 32-bit recording uses decimal points and exponents (power or index – indicating how many times a base number is multiplied by itself). As a result, much larger and smaller numbers can be represented. The dynamic range – the difference between the quietest and loudest sounds that can be recorded – is incredibly wide: approximately 1528dB. This far exceeds the capabilities of fixed point (integer) PCM recording equipment. To put this into context, the dynamic range of a 16-bit WAV file is 96.3dB and a 24-bit WAV file is 144.5dB. Given that the loudest sound ever heard on Earth is estimated to have been around 200dB at source (the Krakatoa eruption of 1883) the format isn't going to have a problem capturing a Formula One race car (around 115dB), a shotgun blast (150–160dB) or a rocket launch (160–170dB). Indeed, the simplest way to think of a 32-bit machine is as a wide dynamic range recorder.

■ ***16/24-bit liner WAV***



■ ***32-bit float WAV***



Clipping advantage of 32-bit float recording.

What does this mean in practice? Unlike traditional 24-bit recording, where signals that exceed 0dBFS result in irreversible clipping, 32-bit float recording preserves the original waveform even if the audio appears to be clipped when viewed in digital audio workstation software (DAW) such as Garage Band, Cubase or Pro Tools. Clip gain can be reduced in post-production, and the audio will sound fine. How so? Remember that 1528dB dynamic range – where 0dBFS for the integer PCM is a hard limit, which cannot be exceeded without distortion – 32-bit offers an additional 770dB headroom.

As well as this clipping advantage for loud sounds, 32-bit offers benefits for recording quiet sounds. With 16- or 24-bit recording, capturing quiet sounds means increasing preamp gain which adds system noise – more or less depending on the quality of your recorder. As no gain is added in the 32-bit scenario, quiet sounds can be boosted in postproduction with no increase in system noise. However, it’s important to note that it won’t stop issues generated by mics, cables or poor technique. Contemporary entry-level 32-bit field

recorders employ dual analogue to digital converters (ADCs) with different gain settings. More expensive professional recorders use multiple ADCs. In the dual converter scenario one captures quiet sounds cleanly, while the other handles loud sounds without distortion. Which ADC is used to process the incoming signal is determined automatically and the system combines these inputs into a single 32-bit float file, preserving detail in both loud and quiet sounds. In the case of the lower-cost units there are no gain controls: they are essentially ‘point and shoot’ devices. What you adjust on the recorder’s screen is the level of the monitor mix, so you can see that something is being recorded. So, in terms of the recorder, there’s not a lot for people to get wrong; none of this ensuring that gain is set between -12 and -6dBFS, choosing file formats, bit depths and so on.

With more fully featured intermediate and professional 32-bit recorders that do allow for gain adjustment what you’re establishing is a zero point. In reality the recorded file is the same wherever you set this – or don’t, as the case may be. Once the file is in a DAW you can simply turn the clip gain up or down and arrive at exactly the same result. However, this isn’t to say that setting gain isn’t important; what will be different when you import the audio will be the appearance of the waveform, so you won’t necessarily need to adjust the gain in postproduction. And this saves time.

Looking at the benefits of 32-bit recording on paper makes it sound like a game changer. In reality there’s a lot of resistance to the technology – much of which is coming from the pro-audio community, and not without good reason. Location sound recordists and post operatives have been working with 24-bit workflows for a long time. This is the current industry standard and it’s the format one group expects to deliver and the other to receive. However, the argument goes beyond just a resistance to change: 32-bit does nothing in relation to poor technique, and microphone and preamp distortion doesn’t just go away. A good location sound mixer will have an understanding of mics, techniques, gain staging and multitrack recording so 32-bit is a solution for managing dynamic range

problems they rarely encounter. Also, 32-bit files are around 33 per cent larger that 24-bit files and this impacts every aspect of recording and production.

So, on the one hand, location sound professionals don’t want to let go of what are considered (rightly) to be good craft skills. On the other, professional post operatives do not want a world where location sound recordists are simply passing more of the workload over to post, and it is certainly the case that 32-bit recording can do this. On more of a technical level there are also debates about the different proprietary methods by which the signals from the high- and low-gain ADCs are blended into a single digital file – and whether this introduces digital artefacts and audible precision errors. And it doesn’t help that much of the marketing for 32-bit sets the technology up as an easy-fix, hassle-free solution for inexperienced sound recordists and videographers.



Zoom F4essentials. Convenient 32-float recording on a budget.

Looked at from the perspective of a field recording enthusiast, developing skills using a 24-bit recorder will stand you in good stead no matter how the technology evolves. However, 32-bit is a contemporary solution that will go a long way to ensuring that your recordings of loud sounds, quiet sounds and environments evidencing a wide dynamic range are captured successfully. For anyone in situations were monitoring and adjusting levels on-the-go isn't practical – journalists in the field, solo filmmakers, people

involved in recording personal engagement in sporting activity, cycling, riding or simply on the move – it offers great promise. And realistically if you're the same person adjusting files in the DAW after a field trip, where the workload sits is of no real consequence.

One final point to note about working with 32-bit files is that you don't have to set the DAW session to 32-bit in order to import them; all contemporary DAWs can read and process 32-bit float files internally – even if they are set to a different bit depth. So they can be treated like any other file. And your software will be able render/bounce files at different sample rates and bit depths (also MP3 if required) so you can supply any format that is required.



Tascam FR-AV2 – offers 24- and 32-bit recording.

Storage Media

With the exception of a few high-end devices with built-in hard disks, all contemporary field recorders use SD memory cards. Some older units still in service (such as the Sound Devices 702 or Fostex FR2LE) use the larger CompactFlash format. SD cards are readily available and even the high-capacity cards are relatively inexpensive. However, all removable media is capable of failing so committing all your audio to a single card doesn't represent best practice: using multiple cards across a field trip helps to spread the risk and backups to computer should be made whenever practical – certainly at the end of every day in the field. Generally speaking, the consensus is to use cards up to 32GB but not higher. Another reason to use smaller cards is that some recorders, such as the older Zoom H4n, take much longer to scan and load large-capacity cards, so there can be a tedious delay at the start of each session. All memory cards need initially formatting in the recorder. It's also best practice to reformat cards once files have been safely backed up elsewhere before reusing them. Not all cards on the market are born equal – for field recording always choose cards with a speed class of 4 or higher. SanDisk, PNY and Kingston cards are generally reliable and long lasting.



Digital recording media: micro SD card in adaptor and older style Compact Flash which is used by older model digital field recorders and digital cameras.

Automatic Level Control (ALC)

Some of the biggest recording challenges present themselves in environments where the ambient sound level is relatively low, but this is interspersed with bursts of occasional high-volume sound – such as the passing of an express train in a rural environment or in sports-audio contexts. The contemporary solution is to consider using 32-bit float recording, but not all equipment offers this. Wouldn't it be good if there were a way for a recorder to automatically turn up and down the recording level in response to the incoming signal? Well, there is; most recording devices have an Automatic Level Control (ALC). On more expensive devices these can be quite subtle but for lower-cost units results can sound unnatural. If the recording contains multiple, periodic loud noises the overall recording can appear to 'pump'. And if someone is recording

an interview or conversation outdoors, the environmental sound almost disappears whenever anyone speaks. The result is very unnatural. I prefer to leave ALC turned off and focus on getting the recording levels right.



Automatic Level Control (ALC): rarely appropriate for field recording, so keeping it set to ‘off’ is the best bet.

FIELD RECORDERS: CONTEMPORARY SOLUTIONS

While the product arena for field recorders isn't massive, there's a good range of alternatives at every price point and the range of devices can be confusing. There are essentially three types of recorders. Firstly, there are basic units with built-in microphones and possibly a mini jack port for an external mic. Secondly, there are the 'Swiss army knife' style units with built-in microphones that also offer XLR/combo/mini jack input connections for external mics. Many offer a multitrack recording option and built-in effects for music makers and can work as a computer interface. To keep costs down, manufacturers have to dance around the cost/quality equation and different degrees of compromise are reflected across the product range. Generally speaking, the more you pay, the less compromises in terms of the audio quality you'll need to put up with. All these units are generally referred to as hand-held devices – something of a misnomer as most can be tripod mounted or secured in a bag while attached to external mics, as required. This is a category of recorder mostly associated with budget to intermediate-level solutions.

However, there is a new generation of higher-specification units that find their way into semi-pro and professional kit bags. This includes recorders that are designed to work with a range of interchangeable, proprietary mic capsules. Thirdly, there are the stand-alone recorders without built-in microphones where all the product development input and budget has gone into delivering the best possible build quality in terms of robust physical containment, metering, preamp/headphone monitoring/limiter circuits, and operational features. They may be able to read and write timecode for image synchronisation and record in 32-bit float WAV format. Looking at the different categories in terms of available products I've included a rough price range – this is for general guidance only; individual prices change but price brackets remain relative.

Factfile 1.3: Anatomy of a Hand-Held Field Recorder

Mic Capsules

Depending on the model these may be fixed, movable or interchangeable. The most common form is what's called an XY stereo array or coincident pair. Here, the mics are arranged with capsules close together, crossing at 90 degrees (sometimes adjustable up to 120/130 degrees for a wider stereo field) to avoid any potential phase issues (*see* Chapter 2). Having the mics close together like this is the simplest solution for creating a trouble-free stereo recording. Some recorders (like the Zoom H4n) let you twist the capsules to give a wider or narrower stereo field. Others, like the Tascam DR 40, allow you to spread the mics into a V formation for a much wider image.

Screen

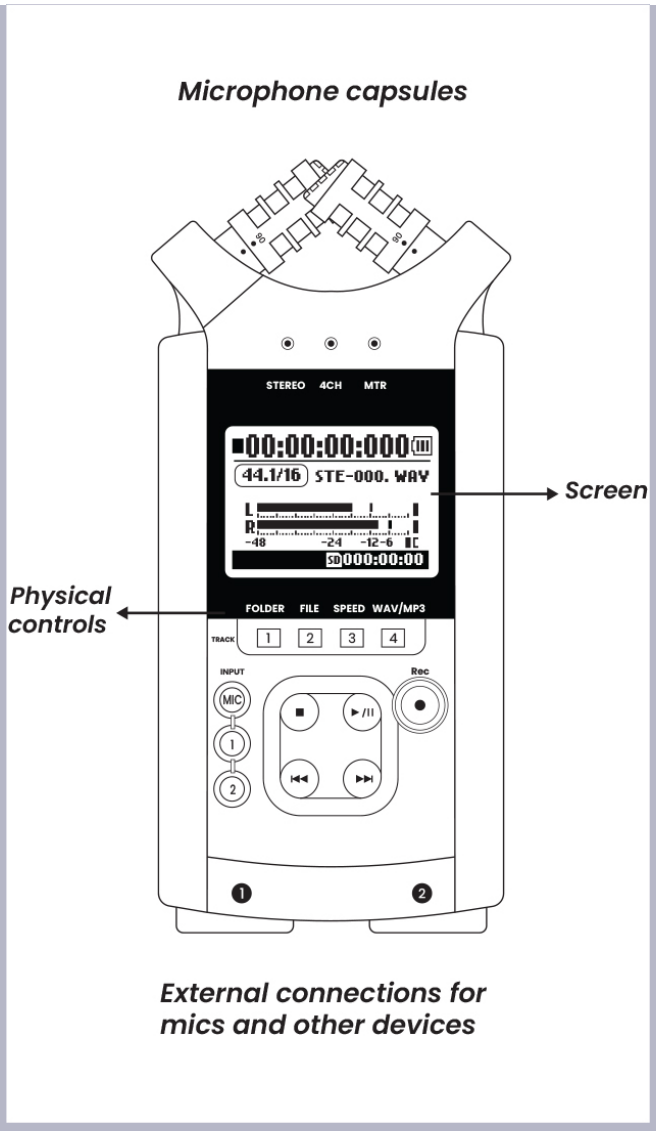
Mechanical dials are long gone. All field recorders have some form of LCD screen with signal level meters and a recording time counter. Touch screens are becoming common. Screens also provide lots of other useful information: file format, sample rate and bit depth, available memory, time remaining, battery health and so on.

Physical Controls

Despite the introduction of touch screens, many recorders still have buttons for transport features such as play, record, fast forward, rewind and stop. Other controls may include record level, headphone level, menu access/navigation, power on and input select (for switching between internal and external mics). Controls will usually be distributed over the top and sides of a unit. Each model is different so you need to familiarise yourself with your kit.

External Connections for Mics and Other Devices

At the budget end of the market, external mic connections are limited to mono or stereo 3.5mm mini jack sockets. Combo sockets are common on intermediate level products. These take both XLR connectors and TS/TRS jack plugs. Professional recorders may have XLR only or Combo connectors so check if you need additional adaptors. There will be a headphone socket, usually 3.5mm. As digital files are mostly transferred directly to a computer the only direct output on lower-cost recorders is the headphone output. If direct connection to a mixer is required, a suitable auxiliary cable can be used. High-end mixer/recorders usually feature dedicated balanced outputs – these are now often on the space-saving TA3 connectors or full-sized XLR outputs. A device such as the Sound Devices Scorpio also offers Dante I/0.



Anatomy of a hand-held digital recorder.

TIMECODE AND WORD CLOCK – IN/OUT

This is only present on units designed for use in film and video production where camera and sound data need to be synchronised.

Field Recorders: Built in Mics + Mini Jack Input (£15–£300)

If you go online and search for ‘low-cost sound recording devices’, you’ll be confronted with a range of products between £15 and £40. Stereo recording is available at even the lowest price point and many feature built-in memory and rechargeable batteries for convenience. However, these units are designed for recording voice memos or lectures and while they can be used for other purposes, the audio quality will be compromised by a number of factors such as low-cost electret microphones and noisy preamps and they are limited to recording lossy formats only. The quality of these dictaphone-style devices gets significantly better once you move beyond £40 and reassuring brand names such as OM System (formerly Olympus) and Philips come into view. However, we are now closing on the price where more flexible, purpose-built devices are available for field recording so unless your budget is severely limited or you simply can’t manage with a phone, there are better options. A budget of £80–£150 will give you access to a good range of serviceable products.



Tascam DR-05X.



Zoom H5.



Sound Devices MixPre-6 Mk II.



Zoom H1essential.



Tascam DR-07X with mics in AB position.

At the lower end of this price bracket, we find basic stereo recorders such as the Tascam DR-05X, DR-07X and the 32-bit Zoom H1essential Handy Recorder. There is also the OM System LS-P1 (marketed as a music and meetings recorder but far more capable than many similar devices) and the LS-P5. With the exception of the LS-P1, which only has 4GB internal memory, all the units use a micro SD card as the recording media, are capable of recording lossless audio formats (as well as MP3) and all feature reasonable-

quality built-in stereo microphones. The OM System models have a third central mic, which is useful where a solid focus is required for a voice or similar sound source, and the LS-P5 is the lowest-cost unit on the market to offer remote control via Bluetooth with the appropriate Android or iOS app. In this group, connectivity for external microphones is very limited: the LSP-1 and 4 only have a mono input socket for external dynamic or self-powered mics. The DR-05X, DR-07X and the H1 support external stereo mics (on a 3.5mm mini jack) and the units can supply plug-in power if required. This is great as it means they can work effectively with a wide range of new low-cost, high-quality electret mics from makes such as LOM and MicBooster, and binaural in-ear solutions from Roland, Soundman OKM and Sound Professionals. What’s the difference between the DR05X and DR07X? With the latter you can move the up-rated mics from the XY position to an AB position to capture a wider stereo field.


MIC / AUX IN
L
R
R-07
Roland
R07 0001.WAV
00:07:07
00:50:02
96k
PHONES
REC
01:22
PEAK
Bluetooth
REHEARSAL
MARK
A B
SCENE
MENU
VOL
INPUT

Roland R-07.



Nagra Mezzo.

If you are looking for a small, high-quality all-in-one solution that can also be used with binaurals or other electret mics, the Roland R-07 is another option. This unit offers wireless connectivity for control, monitoring and listening back via Bluetooth using the appropriate Android or iOS app. This is very useful if you want to leave the device in a tree and retire to a safe distance. Also worth consideration are Nagra's Pico and Mezzo recorders – a bit more expensive but the brand has a reputation for delivering superb audio solutions and excellent build quality. The Pico has 4GB internal memory, built-in stereo mics and a 3.5mm mic input jack that supplies plug-in power. It can record both MP3 and WAV files at 48kHZ/16-bit. The Mezzo has the same fundamental features but offers 8GB internal storage and variable WAV recording options up to 96kHz/24-bit, making it better suited to a range of field recording needs.

The only currently available Sony unit is the PCM A10, which is a linear PCM recorder. While it's certainly no slouch in terms of audio quality, its similar form factor to the lower-cost Zoom and Tascam units such as the DR-07X or H1essential places it in a highly

competitive product arena. However, if you are looking for a pocket-sized, high-quality all-in-one recording solution, the PCM A10 is a serious contender.



Zoom H2n in 4-channel surround recording mode.

Also in this category we find a Zoom recorder with quite a unique feature set. The H2n Handy Recorder has been around since 2007 and has been revised several times – with the 32-bit H2essential

version arriving in 2025. Note however that this updated version offers mono, stereo and mid/side stereo (*see* Chapter 2) recording options, but not 4-channel surround – for that, Zoom now has the H3-VR. The convenient size and surprisingly varied recording options of the standard H2n have made it a firm favourite with amateur users, students and sound artists. It's also a gateway device for people interested in exploring multichannel audio. I'll be covering Ambisonics and multichannel sound in detail in Chapter 4 but given the price point of this unit, it seems appropriate to introduce it here. The current H2n looks quite a bit like a large capsule condenser microphone and indeed it can be used as a multi-purpose USB microphone for voices and instruments. However, the enclosure contains 5 mics that enable the unit to offer four different recording modes. As well as the standard X/Y stereo recording it offers mid/side recording (a technique that allows you to adjust the width of the stereo image after it has been recorded – *see* Chapters 2 and 6 for more details), 2-channel surround and 4-channel surround. In Spatial Audio mode, the unit records a 4-channel AmbiX WAV file that includes omni, left/right, and forward/backward information (*see* Chapter 4).

Factfile 1.4: Field Recording with a Mobile Phone

If you're new to the idea of field recording, and find the idea of buying dedicated equipment daunting, the good news is that almost everyone now carries a capable recording device. On its own, a contemporary smart phone isn't going to give you the exceptional results you'd expect with a dedicated recorder or specialist microphones, but it will do a remarkably good job. Many people don't realise that phones can record in stereo because the default setting is mono to keep file size down. They can also record high-quality lossless audio (WAV files) if settings are adjusted. To get the best results download a third-party app such as RØDE Reporter (iOS and Android); your default voice recorder app may have limited options. However, even the native app will be capable of capturing environmental sound in good detail.



For best recording results with your phone, use wind protection and place the phone on a surface rather than hold it in your hand.

Where is the mic? Most phones have at least two, with one near the camera and one on the bottom edge for the voice. These mics are all omnidirectional, so you don't need to point the phone in a specific direction. If possible, don't hold the phone while recording, as this will generate handling noise and potentially cover a microphone. Take it out of the case if that partially covers a microphone. Recording outdoors? Even in relatively still conditions you're likely to get some buffeting noise from wind hitting the microphones. Either try to find a sheltered spot or use a suitable sized windjammer – the type made for recorders and microphones by companies such as Rycote, RØDE and MOVO. Rycote also make small stick-on micro windjammers that can be used with your phone. Alternatively, explore the creative use of knitted items such as scarves and gloves.

If you're interested in going further with phone recordings, there are add-on mics specifically for the purpose. For older

model iPhones there is the RØDE iXY, a high-quality stereo mic. However, this has now been discontinued. Zoom has the iQ6 and iQ7 MS. All these options, which cost between £100–£160, have internal A/D converters and the audio signal is sent to the smartphone digitally via the Lightning or USB-C connector. Zoom also offers a stereo mic for Android phones, the Am7. Using one of these microphones with your phone will give you great results, similar to those achievable with a dedicated hand-held recording device.

Field Recorders: Built in Mics + XLR/Combo Inputs (£170–£450)

There’s plenty of choice and price variation in this category and the trick is to identify what your needs are before you start shopping, especially where units with different features are clustered around the same price point. All the devices we’ll look at here have either built-in microphones or interchangeable, proprietary mic capsules and XLR/combo connectors for external mics. They can record WAV and MP3 files and all can be used as an audio interface with a computer. And some feature a small internal speaker for reference monitoring.



Tascam DR-40x showing both potential on-board mic positions and enabled for 4-channel recording using additional external mics.



Zoom H4n: recently discontinued, but many remain in service giving excellent results.



Zoom H4essential 32-bit only recorder.

Tascam's DR40x and the now discontinued Zoom H4n Pro have been best sellers at the lower end of this price bracket for many years. These are units I encounter on an almost daily basis; they're a staple of college and university music technology provision, popular with adult learners in my field recording classes and often a first purchase for visual artists looking to incorporate, or improve, the quality of sound in their work. Both have built-in microphones and combo jack inputs for external mics, they can record 2- or 4-track

audio (mostly used for making multitrack music demos) and both can also be used as a USB audio interface with a computer or other device. The 32-bit H4essential replaced the H4n Pro in early 2025, and the unit offers essentially the same feature set as the old model. In addition, the Essential series is the first designed with accessibility in mind for the visually impaired: you can navigate the menu with audible descriptions through the built-in speaker or headphones. The language options are English, Spanish, French, Japanese, German, Italian and Chinese. Another advantage these units have over the DR-40x is the stereo 3.5mm input that supports plug-in power. Inexplicably, the DR-40x doesn't have this – despite the facility now being available on the new DR-05 and DR-07.

Electret mics, such as binaurals, need plug-in power (a 3–5 volt supply) to operate. You can use these mics with the DR-40x but you will need an additional preamp (such as the Sound Man A3) or adaptors that convert phantom power to plug-in power (such as the Deity D-XLR Phantom Power Converter or Audix APS910). The DR-40x also has some advantages over the Zoom, including its conventional approach to digital recording for those not quite ready to put all their eggs in the 32-bit basket! With the DR-40x you can also change the position of the built-in mics from XY to AB, which delivers a much wider stereo field if required. It also has a dual record function that generates a simultaneous second copy of the audio at a lower level – so a failsafe in case the primary take overloads and distorts. And its menus and controls are very easy to navigate.


ZOOM
L
R
001:23:45
XY5
FOLDER01
ZOOM0001
1
2
L
R
1
2
H5
Handy Recorder
ZOOM

Zoom H5.



Zoom H6essential.

In the upper half of this price bracket, we encounter more Zoom and Tascam products. Looking at Zoom's H5, the new H6essential (which has replaced the older H6 model) and H8 recorders for the first time can be confusing as prices across the range are not so different. The H6essential is a 32-bit recorder, which is one of the more obvious differentiating factors. It's probably easiest to consider the H5 as a logical alternative to the discontinued H4n if you want a conventional digital recorder. The H6essential and H8 make a

slightly different offer – principally because of the number of extra inputs they have. This makes them suited to a range of additional tasks beyond basic field recording, including multichannel and Ambisonic recording (*see* Chapter 4). One thing all three units share in common is the ability to use Zoom's expanding range of interchangeable microphone capsules. No, you won't get the same degree of audio excellence that top-end external mics will deliver but these Zoom capsules perform extremely well, and your budget will be in the £100s rather than £1000s.



Zoom XAH-8 cartridge with mics in AB position.



Zoom SGH-6 mono shotgun cartridge.

The H5, H6essential and H8 all ship with an XY stereo capsule – the H5 with the XY-H5, the H6essential with the XYH-6e and H8 with the XY-H6. These capsules are interchangeable and have similar sonic characteristics. There are differences in form factor: the XY-H5 mics are ‘shock mounted’ using a flexible ring and fixed at 90 degrees. The XY-H6 capsule isn’t shock mounted and doesn’t have the level protection bar. However, it’s adjustable between 90 and 120 degrees for a wider stereo capture. XYH-6e has the angular product

styling of all the Essential series units. Note that the XY-H5 and the XY-H6 feature a mic/line input jack on the capsule that can be used to connect an external mic or line-level device. This socket provides plug-in power for electret mics such as lavalieres and binaurals. Unfortunately, this feature isn't to be found on the H6essential: you will need the alternative EXH-6e Dual XLR/TRS Input Capsule. There is another Zoom stereo capsule, the XAH-8, which is adjustable to offer both XY and AB mic positions. In addition, you can purchase the SGH-6 mono shotgun capsule; the SSH-6 mid/sides stereo shotgun capsule; the MSH-6 mid/sides capsule; the XEH-6 module (which replaces the mics with 2 extra inputs for external mics – this doesn't provide phantom power however) and the VRH-8 Ambisonic capsule. And the range continues to expand rapidly.

Turning to look at the individual recording units in closer detail, the H5 is the smallest and lightest of the three. If you can't envisage a situation where you'll need to use more than 4 microphones at the same time (Stereo XY or other capsule and two externals), the H5 is one of the obvious options as a first or second machine. The H6essential has 4 inputs for externals (6 with the optional XEH-6 capsule) and is likely to appeal to those interested in podcasting or small-scale music multitracking projects as well as field recording. The H8 is something of a multi-purpose behemoth – when fully cabled up with its 6 external inputs (plus 4 additional inputs with the optional EXH-8 module – which can provide phantom power) you're potentially looking at full band recordings, multi-mic video/film shoots (no timecode in/out option but it does have a sound marker function that is useful for synchronising image and audio) and other multichannel uses. The H8's operating system offers three distinct start-up modes: Field app, Podcast app and Music app. The H6essential can be operated remotely using the BTA-1 wireless adaptor and the dedicated H6essential Control app for smartphone or tablet. The H8 has a similar feature. With the H6essential the BTA-1 adaptor is also needed if you want to input timecode, via Bluetooth from an UltraSync Blue (made by Timecode Systems).

Note that all three devices have quite different screens. The H5 has a fairly basic backlit LED screen similar to the old H4n Pro. The H6essential has the new colour screen of the Essentials series. On this model it's tilted at a slight angle, which makes the unit more practical for attachment to a DSLR camera. The H8 has a full colour touch screen, which makes the different operation modes very practical – for example, using on-screen faders for channel inputs.

At the upper end of the price bracket (around £340) Zoom has introduced the new H5 Studio model. While not aimed specifically at field recordists, it will no doubt perform extremely well. Billed as a professional grade product offering 6-channel 32-bit float recording (but with the addition of dedicated gain knobs for those who still want to manage the audio input levels), F series studio-grade preamps and a new microphone capsule. It also has enhanced RF shielding to help exclude interference from smartphones, Wi-Fi routers and other sources. With the TCA-1 timecode adapter it can also generate and output timecode or work with an external clock.

Moving over to the top end of Tascam's range of hand-held recorders, we find the Portacapture F6 and F8 (the DR-44 was discontinued in 2024). The main difference between the two is the number of mic inputs and the corresponding number of tracks they can record. The X6 has 2 extra inputs on XLR connectors and can record 6 tracks in total (4 independent tracks + stereo mix of the 6) and the X8 up to 8 tracks (6 independent tracks + stereo mix of the 6). All the mic inputs on the X8 are on combo connectors so there is more flexibility for using mics and devices with TS/TRS plugs (you can do this with the X6 but you would need adaptors). There are other more subtle differences: the X8 has a bigger touch screen, the built-in mics of the X6 swivel but aren't detachable whereas those on the X8 have a larger, uprated capsule design and can be taken off (useful if you're just using external mics); both offer 32-bit float recording but the X8 offers the higher 192kHz sample rate while X6 offers a maximum of 96kHz. Overall, the F6 is the more compact unit (lighter by 100g) and the F8 is more fully featured with extra

connectivity which accounts for the roughly £100 price differential between the two.



Zoom H8 – screen showing different app operating modes.



Tascam Portacapture F6.



Tascam Portacapture F8 showing both potential mic positions.

As with the Zoom H8, the introduction of a touch screen facilitates the move to an app-based approach to operation for both Portacapture models: there are six to choose from. The F6/F8 are the only products on the market offering a setting specifically aimed at Autonomous Sensory Meridian Response (ASMR) enthusiasts. Altogether we have apps for ASMR, Voice, Music, Manual (the basic app of the unit), Field and Podcast. There is also provision for

remote control – this time via Bluetooth using the optional AK-BT1 adaptor and the controller app for iOS and Android devices.

Field Recorders: Stand-Alone Solutions, No Built-In Mics (£150–£13,000)

Until a few years ago recorders of this format were all higher-cost units aimed at the upper end of the consumer and professional markets. However the introduction of the Zoom F1-SP changed that. This is a 2-track recorder that's designed to be used with the proprietary Zoom capsules for the H5 and H6 and electrets that require plug-in power via the 3.5mm jack socket. It's small, light and ships with the SGH-6 mono shotgun capsule and a windshield for around the £200 mark. And there's a camera shock mount if required.



Zoom F1-SP – complete with shotgun capsule, foam windshield and camera mount



Zoom H1-XLR Handy Recorder – another 32-bit float stereo recording option. Designed for external mics only.

If you don’t feel the need for a recorder with built-in mics and are happy with a unit that only offers 32-bit float recording, the Zoom H1-XLR is a high-quality low-cost solution at around £140. You can also connect it direct to your mobile phone via USBc to deliver a massive improvement in video sound quality. And it works as a 32-bit audio interface for the home studio.



Zoom F3.

For an extra £100, the Zoom F3 is another stereo 2-track recorder with XLR connections and a range of more advanced features. It can be used as a computer interface and can be controlled from a smartphone or tablet using the BTA-1 wireless adaptor. It can also receive timecode from an UltraSync BLUE (made by Timecode Systems). It's another 32-bit only recorder (44.1kHz, 48kHz, 88.2kHz, 96kHz, 192kHz) and has a much-simplified operating system. The unit is clearly designed to appeal to

the budget filmmaker, but it’s also a practical tool for the field recordist who’s looking to achieve hassle-free results.



Tascam FR-AV2.

Tascam also offers a small format audio recorder that fits neatly into this group: the FR-AV2. At around £340 it steps up the price, but this unit offers 24-bit as well as 32-bit recording and has enhanced timecode functionality and a range of other useful on-board features such as a low-cut filter, compressor, limiter and noise gate.


ZOOM
MultiTrack Field Recorder
F6
190101_001
48.00
LINE
1 2 3 4 5 6 L R
INT 29.9D
00:10:42:06
PFL
1
2
3
4
5
6

Zoom F6.



Zoom F8n.

For those who want the option of 16-bit, 24-bit and 32-bit recording, Zoom's solution in this price bracket is the F6. With 6 XLR inputs and time code in/out it can be employed in a wide range of situations: basic field recording, multichannel and Ambisonic recording and for sync sound with video/film audio. Need even more inputs? Zoom's flagship audio recorder the F8n Pro has 8. It also features a socket for connecting Zoom's propriety mic capsules. However, while the F8n is certainly a portable recorder, it seems highly unlikely that you'd purchase it with a view to using these as your main recording mics.

Which brings us to the Sound Devices range of products. Sound Devices has been servicing the needs of film professionals and field recordists since the mid-1990s and its recorders and field mixers have been used on dozens of major motion pictures and won numerous technical awards. The 702 (or the 702T with timecode) was a highly desirable machine; excellent build quality, great sounding preamps and designed to run on Sony-mount L-type Li-ion rechargeable batteries – used by many cameras – making it a really great-sounding, reliable and practical piece of equipment. Second-hand units still command a very good price today. The company's latest machines have maintained a firm hold on the professional arena. And again, the contemporary MixPre-3 Mk II, MixPre-6 Mk II

and MixPre-10 Mk II are all excellent machines, each with similar sonic characteristics but offering more recording tracks and inputs as you ascend through the range.



MixPre-3 Mk II, MixPre-6 Mk II and MixPre-10 Mk II: more or less recording tracks depending on your needs and budget.

Global economics of the last decade haven't been kind to purchasers of American-made equipment in the UK and Europe, and the days of a Sound Devices unit for under £1,000 are long gone. That said, the price difference between the MixPre-3 and 6 seems to have been squeezed to the point where not opting for the 6 seems churlish – given that the presence of the fourth XLR connector opens up the possibility of exploring multichannel recording and Ambisonics. One point of confusion that's worth clearing up here is that while the MixPre-3 II has 3 XLR mic inputs, the MixPre-6 only has 4. And likewise, the MixPre-10 has 8. However, the MixPre-3 has 5 inputs (3 mic-line inputs, 2 on Aux In), the MixPre-6 has 6 (4 mic-line inputs, 2 on Aux In) and the MixPre-10 has 10 (8 mic-line inputs, 2 on Aux In). It's probably easiest to think of the MixPre-3 as a 5-track recorder (stereo mix + 3 isolated tracks), the MixPre-6 as

an 8-track recorder (stereo mix + 6 ISOs) and the MixPre-10 as a 12-track recorder (Stereo mix + 10 ISOs). What does your investment get you? All the good stuff: 44.1kHz, 48kHz (also 47.952kHz 48.048kHz – it's a film thing…), 96kHz, 192kHz sampling frequencies and 16, 24, 32 bit-float recording; high-quality, low self-noise Kashmir mic preamps; dedicated, manual mic gain controls with built-in record monitor/overload lights; high-quality headphone amp; a bright, clear mini touch screen; XLR inputs (combo XLR/TRS on the MixPre-6 and 10), a 3.5 aux input supporting plug-in power; timecode in/out; Bluetooth connectivity for remote operation using Sound Devices' Wingman app for iOS or Android and they can be used as a USB computer interface. They can be powered by AA batteries (four or eight) or an optional L-mount sled for Li-ion batteries.



Sound Devices 833: professional quality mixer/recorder.



Sound Devices Scorpio: high-end audio field recorder designed principally with film-sound needs in mind.

If the Sound Devices MixPre Series doesn’t adequately fit the bill in terms of sonic excellence, adequate connectivity and routing flexibility, the company also makes its 8-Series mixer/recorders: the 833 (6 mic/line preamplifiers, 8 channels, 12 tracks), 888 (8 mic/line preamplifiers, 16 channels, 20 tracks) and the Scorpio (16 microphone preamplifiers, 32 channels, 36 tracks). However, these are designed with film and TV needs in mind and with prices between £7000–£13,000 will be somewhat over specified for the needs of most field recordists.



Nagra VII: high-quality professional recorder from historic brand.

At this upper end of the market, we also come to Nagra. This iconic brand has been associated with film sound and field recording since the mid-1950s. Its analogue tape machines were ground-breaking products known for both engineering and sonic excellence and the company has continued to develop digital solutions for today's professionals. The company's hand-held Pico and Mezzo recorders that we mentioned earlier in this chapter are a rare nod towards a wider market. Today, Nagra's principle digital recorder is the VII. Unfortunately, the flagship 8-channel VI went out of production in 2018. One presumes an update is in the wings. The VII exudes Nagra classic style: understated, iconic and practical. It's a 2-channel 16/24-bit digital recorder with a colour touch screen offering sampling frequencies up to 192kHz, with inputs for 2 mics plus AES 42 for digital microphones. The design is essentially modular: the series offers a selection of internal circuit cards to allow it to be adapted according to requirements. The hardware options consist of 3G and Wi-Fi, SMPTE/EBU timecode and ISDN communication possibilities.

For the sake of completeness at the top end of the audio recorder market, it's worth mentioning the AAton product range. Unfortunately the company was declared bankrupt in 2024 and no new products are available. However, units such as the 16-channel Cantar Mini

(Chris Watson’s current machine of choice) and 24-channel Cantar X3 are still very much in use in the professional arena.

Factfile 1.5: First Generation Digital File Recorders

While not strictly legacy technology, newer models have superseded the machines that started to appear in the early 2000s. Older devices include the Fostex FR2-LE and the Sound Devices 702. If in good working order, these units still sound good and transferring files to your computer is quick and easy. They either use standard SD cards or the older Compact Flash format, which is still readily available. With few moving parts they tend to be very reliable. While second-hand Sound Devices units tend to sell at a premium, other units can often be picked up at very good prices. You can bag a second-hand FR2-LE for around £80: robust design, great sounding preamps – what's not to like!



Fostex FR2 LE. A real workhorse machine from the early 2000s that can still keep up with modern machines for sound quality and flexibility.

MONITORING IN THE FIELD: HEADPHONES

While a rolling counter and flicking level meter indicate that recording is taking place, it doesn't tell the whole story. RF interference? A bad cable causing crackle every time you move? An intermittent connection? An issue with the microphone capsule? Inadequate wind protection? And then there are the more 'artistic' issues to consider where some close listening indicates that a change of microphone or location will better reflect the content you are attempting to capture. Studio engineers talk of 'sweet spots' – precise microphone locations where the sound of an instrument sounds just right. The best way to find these is to move the mic while listening on headphones and stop at the point where the sound quality is what you want – and that's where you set up the mic.



Anna Borzello monitoring in the field. Closed-backed headphones enable you to listen to the sound being recorded via the microphones in isolation from the everyday listening experience.

It’s the same in the field: if you treat monitoring as a creative act it will help ensure great results. Sometimes, simply moving a mic a matter of centimetres to the left, right, up or down can make the difference between an average and a great recording. Is the recording too reverberant or is there an annoying echo that seems so much more obvious on the recording than it did at the time? None of these things is apparent if you simply press record and hope for the best. Microphones are all different and none ‘hear’ exactly like your ears, so learning to listen through your mics is an important, and often overlooked, craft skill for the field recordist.

Headphone monitoring in the field poses several problems. Ideally, you want to hear what you are recording at a realistic volume level while excluding the actual sound around you. This requires a good level of sound isolation. Pushing your monitoring level to mask the sound bleeding in from the environment in order to hear detail

can be highly fatiguing. It's also likely to make you aware of additional system noise from the headphone amp. While this is purely part of the monitoring chain and not being recorded, it's hardly a comfortable or reassuring experience. Monitoring too loud is also a long-term threat to your hearing.

Standard ear buds aren't suitable for field monitoring as they are designed to make music sound good and the tonal characteristics are often tweaked to deliver 'impressive' bass results. What the field recordist needs is a good pair of closed-back, low-to-medium impedance (so you don't have to crank the headphone amp to get a decent level – *see* Factfile 1.6 for more detail) over-the ear headphones that deliver what's called a flat frequency response. This means they have a neutral sound profile without any artificial enhancement so what you hear is as close to reality as possible.

Thankfully, there are quite a few options, especially as many of the headphones used by musicians in recording studios need to exhibit the same characteristics (the closed-back design also stops sound bleeding out of the headphones to be picked up by the microphones in the live room). A budget of between £60–£150 will enable you to purchase a great pair from a range of brands such as AKG, Sennheiser, Audio Technica, Sony, Beyer Dynamic or RØDE. As well as improved sound quality, many of the higher-cost options offer versions with a range of impedances and detachable/replacement cables – which is both practical and, as the cable is always a weak point, a potential long-term cost saving.



Audio Technica ATH M50x headphones.



Beyer Dynamic DT 770 Pro headphones.



RØDE NTH-100.

Comfort is another significant issue when choosing headphones, especially if they double in the field and the home/studio. Note however that open-backed headphones are preferable in the listening/mixing environment – more of this in Chapter 6. Head sizes vary and so does the form factor of the different brands so if you can try before you buy, all the better. As a student I used a pair of Sony MDR-750S headphones and then stayed with them for a good while as I was used to their characteristics. Audio Technica's ATH M50x

and Beyer Dynamic DT 770 Pros are both popular with amateur and field recording professionals. RØDE has the NTH-100, which has unique styling and removable memory foam earpads for extended wear comfort.



Nagra Model 1 Field Monitoring Headphones.

While I've focused here on brands and models at a lower, but practical, price point, there are many options for closed-back headphones at higher prices if you have a bigger budget. For example, Nagra (in association with Audeze) have their own Model 1 Field Monitoring Headphones which are advertised as specifically designed to complement their own recorders – but would no doubt be a superb monitoring solution in any setup.

If you want an in-ear monitoring (IEM) solution the market is quite crowded because musicians now use these for live performance. Check out brands such as Shure, Stagg, Behringer, Sennheiser, Linsoul, InEar, Fischer Amps, Hörluchs, AKG and 64 Audio. Prices range between £40–£3000(!). Again, a budget of £60–£150 will give you plenty of choice.

FACTFILE 1.6: HEADPHONES – WHICH IMPEDANCE?

The basic rule of thumb is that the higher the impedance, the more power/voltage required to get a good sound level. For phones and mobile devices, low impedance (35–40 Ohms and below) is best. For audio interfaces, which have more powerful headphone amps, the medium option (80–100 Ohms) is what's needed; with low impedance headphones the monitor control will have very little flexibility and hover close to the zero mark. With high impedance headphones the control will probably need driving close to maximum, again offering little flexibility. Professional and broadcast systems, especially in the early days, tended to be much more highly powered which is why high impedance headphones were introduced – to stop potential hearing damage. Do higher or lower impedance headphones sound better? This is a hard one to answer because so many design factors come into consideration, as well as price. If you're only going to use headphones with a field recorder, low impedance is ideal. If you want your headphones to also serve in the home/studio environment, choose medium.

Do I Always Need to Monitor?

While I've stressed the importance of monitoring in the field as both a necessary and creative part of recording, it's important to note that it's not always the case. Sitting by your rig obsessively monitoring for hours on end will do no one any favours and may not deliver the best results; there are times when removing yourself entirely from the environment using long mic cables is the best approach – especially when recording wildlife. And there are times when leaving a drop rig to capture what takes place in your absence is the best solution.



Monitoring in the field: there are times to step away from your equipment and trust the environment to deliver.

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2
CAPTURING AIRBORNE SOUND

While the pursuit of interesting and unique field recordings can lead us to explore the transmission of sound through water, solid materials and complex structures and to investigate the electromagnetic realm, the successful capture of airborne sound is the basis of field recording practice. In this chapter, we'll focus on the knowledge, techniques and equipment that will enable you to get excellent results every time. As with most things in life, practice and experience count for a great deal, but the basics are relatively straightforward. However, before we start to discuss different types of microphones, let's briefly look at human hearing – not specifically in terms of its mechanics or the ways in which we identify content, but how we unpack spatial cues and other psychoacoustic phenomena as meaningful data. Then, the tools and techniques for capture and reproduction become much easier to understand.



Recording coastal sound: MicBooster Pluggy XLR Primo EM272Z1 omni pair.

HUMAN HEARING

Humans, like other mammals, have two ears which are essentially omnidirectional microphones. These are positioned on either side of our head – this creates something called the head shadow. The ear that is shadowed by the head receives sound a few milliseconds later than the ear closest to sound and the amplitude is fractionally reduced. The signal is also subject to some tonal colouration. The brain interprets the difference in the sound heard by the two ears as directional information. In addition, the size, shape and density of the head, external pinnae, nasal and oral cavities and upper body all contribute to the way we localise sound – enabling us to sense distance, direction and whether sounds are coming from below or above. Overall, the mathematical model created for each ear by the summation of the data is referred to as its head-related transfer function (HRTF). Everyone’s HRTFs are different and the ability to localise sound can be severely compromised by genetic or age-related hearing loss, aural diversity or trauma.



Human hearing – the shape and size of the head and pinnae contribute to a unique hearing profile.

In terms of frequency range, human hearing is relatively limited when compared with many other mammals. Generally speaking, the range is between 20Hz and 20kHz. Dogs have a range that extends from roughly 40Hz to 60kHz and for bats it can be an extraordinary 10Hz and 120kHz. Hence, many species are very aware of, and use, what we call ultrasound: sound outside the range of human hearing. Ultrasound is a fascinating realm that I'll be investigating in the next chapter.

Also note that humans naturally lose the top end of the frequency range though aging. This starts remarkably early. Few 18-year-olds can still hear above 17.4kHz and by the age of 40, the upper limit for most people is around 15kHz. By 50, for many people the upper limit will be around 12kHz. So look after your hearing as best you can; any additional damage will only exacerbate this natural decline. This all goes to show that you can't make any assumptions about what

others can or can’t hear – which means that concepts of ‘hi-fidelity’ are extremely relative in terms of high-frequency content.

MICROPHONE PLACEMENT

Where do I put the microphone? This depends on whether you want to capture a specific focus or a general impression of the sound field – or where in the balance you want to establish the relationship between specific sounds and the overall background. When a mic is close to a sounding object it requires the minimum preamp gain to get a good level and this minimises interference from ambient sound. As the distance from the source increases, we need to turn up the gain to maintain a good recording level (32-bit recording considerations aside) and the level of ambient sound captured increases.

There is another way to look at this: if there are no obstructions, sound waves propagate outward from a source in a spherical pattern. As sound moves away from the source the total amount of energy remains the same, however this gets distributed over a wider area. Eventually, it becomes so dispersed as to be indistinguishable from the general environmental sound. Beyond a certain distance, sometimes referred to as the sound's acoustic horizon, it will disappear entirely. Doubling the distance from the mic to the sound source (an increase of radius) means the area will quadruple. This reduces the apparent volume of the source sound by roughly one half. In terms of acoustic energy this equates to a loss of about -6dB SPL. This is used as a general rule for each doubling of distance between mic and source. This is a useful observation, but something of a blunt instrument in reality – based on idealised conditions where there are no impediments to sound transmission or other factors such as wind speed and direction. Obstructions (and related reflections) and conditions impart specific losses (and gains) and these in turn are frequency dependent.

Get Close to Your Subject

Chris Watson points out that the absolute key issue with location recording is where you put the microphones; the craft of getting them to the right place is most of the work. Where you put your ears to listen before you set up and start recording comes first. It's not always the case that you need to have the most super directional microphone. Shotgun mics, which we discuss in detail later, have their uses, but standing 50m from your target is not going to deliver intimate detail. Watson prefers to use omnidirectional mics about 3m away (or closer if possible) from the source of the sound. Then, depending on the subject, it's a case of running long cables to get well away and record a very close perspective.

If we reflect back on the techniques of a field recording pioneer such as Ludwig Koch there are still lessons to be learned. After spending days in the field to locate the best perches for singers he, or an assistant, would be up a tree or pole installing a microphone. They would often install multiple mics and mics in surrounding trees – not with the intention of mixing or blending sound, but to increase the likelihood that a bird would perch near one and sing. Back in the recording studio Koch would monitor the incoming mic channels and send the one with the best response direct to the cutting lathe. Another exponent of close mic'ing we can learn from is Victor C. Lewis. His recordings of birdsong from the 1960s evidence fantastic detail – achieved through endless observation and careful mic placement at close proximity. Close mic'ing today still delivers the same benefits and the existence of multichannel recorders makes location multi-mic'ing a relatively stress-free process.

Chris Watson has very successfully navigated a path between creative artist and professional field-sound recordist. Indeed, he is one of the few location sound professionals with a public profile – largely because of his television work which includes many programmes in the David Attenborough *Life* series including *The Life of Birds* which won a BAFTA Award for Best Factual Sound in 1996, and as the location sound recordist for the BBC series *Frozen Planet* which also won a BAFTA Award for Best Factual Sound (2012).



Chris Watson recording on location in Arizona using AAton Cantar recorder. (Photo: Dr Patrick Avery)

Watson's parents gave him a portable National reel-to-reel recorder at the age of 11. As a teenager in the 1970s he discovered the book *Composing With Tape Recorders* by

Terrence Dwyer and was soon recording household sounds (doors squeaking, the fridge, the budgerigar…) slowing them down and playing with the sounds. In 1973 along with Richard Kirk and Stephen Mallinder, he formed the experimental group Cabaret Voltaire. In the early 1980s he went to work for Tyne Tees Television, receiving on-the-job training in the station's sound department. With a long-standing interest in location sound, he began to explore Northumberland. After Tyne Tees he went freelance and has developed a career spanning film, television and radio, sound installation, solo performance and collaboration with numerous artists and releases on the Touch label.

As a freelance composer and sound recordist, Watson specialises in creating spatial sound installations that feature a strong sense and spirit of place. In 2013 he received a Paul Hamlyn Composers Award and international galleries and festivals such as Sheffield Millennium Gallery, Opera North in Leeds, The National Gallery and The Royal Opera House in London, The Louvre in Paris, the Aichi Triennial in Japan and Dark MoFo in Tasmania have commissioned his installations.

DIFFERENT TYPES OF MICROPHONES

Condenser Microphones

True Condensers

The majority of microphones used for field recording purposes are condensers. Broadly speaking there are two types – 'true' (also called DC-biased) condenser mics and electret (also called back-electret) mics. Both function via the same basic principles: the microphone capsule contains a thin, flexible diaphragm and a fixed backplate that functions as two sides of a capacitor. When a sound wave moves the diaphragm, it changes the capacitance of the circuit, creating a small, variable electrical voltage – and this is the signal that is sent to the recording device.



Small diaphragm condenser microphone: AKG C451B. Small condensers deliver very accurate results.

True condensers come in two basic formats, featuring either a small or large capsule design and both require phantom power – *see*

Factfile 2.1. The small capsule design is the most commonly used for field recording. The mics themselves are generally compact and the low mass of the diaphragm means that inertia has little effect, so they respond quickly to both incoming impulses and the cessation of sound. Hence, they provide a very accurate rendition of the sound waves. The polar/pick-up pattern (*see* Factfile 2.2) is very consistent, and this is what you want when recording natural or built environments, large-scale events and distant sound sources. Small diaphragm condensers handle higher Sound Pressure Levels (SPL) well because of the greater distance between the diaphragm and backplate, so there is less chance of distortion. Their frequency response is very good in terms of both high and low sounds, making them ideal for recording detail. This extended frequency response results in a very accurate, 'natural' sound which is tonally true to the original source – as a result they are often referred to as having a relatively flat or neutral sound profile.

Large capsule condenser microphones, common in studios, offer excellent signal-to-noise performance and are ideal for capturing quiet sounds close up. Though less rugged and moisture-sensitive, they deliver warm, characterful recordings. Their slower response and uneven off-axis pickup make them less suited for ambient or event recording but perfect for dialogue, animal/pet sounds, ASMR and detailed sound work.

Electret Condensers

Electret microphones are a type of condenser microphone that uses a permanently polarised material, known as an electret, as the diaphragm. They also require additional power to operate, generally between a 3 and 5V supply, which is called plug-in power. This is only provided on the 3.5mm mini-jack mic input sockets like the ones you'll find on many Zoom recorders or on phones and laptops. For recorders that don't supply it you can buy dedicated preamps (like the Soundman A3) or an external phantom-to-plug-in converter (like the Soundman PPA) or individual plug adaptors such as the RØDE VXLR+ 3.5mm TRS to XLR Adaptor or Deity D-XLR 3.5mm. Some

electrets, like the FEL MicBooster Pluggy XLR Primo EM272Z1 can convert phantom power to plug-in directly.



Large diaphragm condenser microphone: Neumann TLM102. More of a studio mic but can be used in the field.



MicBooster Pluggy XLR Primo EM272Z1 omni pair. Low cost, great sounding mics suitable for all environments.

Electrets are everywhere – and the quality varies immensely. Low-cost units are used in phones, laptops and sound toys and high-specification versions are used as measurement mics. All the built-in mics you'll find on hand-held recorders are electret condensers. Most binaural in-ear mics use electret capsules. In recent years companies like LOM and FEL MicBooster have introduced high-quality, low-cost electrets in various innovative formats. Indeed, these new tools are helping to drive a new wave of interest in field recording because quality solutions are now available to beginners and amateurs. And many professionals are adopting the same tools as they deliver excellent results at a 'risk-able' price – so are ideal for use in hazardous situations, long-term installations or multichannel recording where many mics of similar specification may be needed at the same time.

FACTFILE 2.1: PHANTOM POWER

Both small and large capsule condenser microphones require power to polarise the diaphragm and power preamp circuitry and impedance converters. Phantom power is a clever solution that bypasses the need for batteries or external power supplies because the DC voltage is sent through the microphone cable. It's called phantom because it does not interfere with the audio signal. All field recorders that accept external mics can supply 24V–48V phantom power – some mics actually need less, but mics will only draw as much current as they need. Accidentally sending phantom power to a dynamic microphone won't harm it, it just doesn't need it. However, never send phantom to a ribbon mic (unless it is clearly identified as an active ribbon like the Golden Age R1) as this can potentially damage the mic.

Dynamic Microphones

While most of the microphones used by field recordists are condenser mics, dynamic microphones have their uses. There are many different makes, but all feature a diaphragm that's attached to a voice coil. This is suspended inside a permanent magnet. When sound waves enter the microphone, they cause the diaphragm to vibrate, which in turn moves the coil, generating an electric current. Dynamic mics are relatively affordable and can handle high SPL without distortion. As a result, they are the mics you see singers using at gigs and they get used for close mic'ing amps and drums. Dynamic mics can be useful in the field when you have very loud sounds to record. For example, for the (1968) recordings for *Memories of Steam*, Kenneth Granville Attwood used pairs of Sennheiser MD 421 or AKG D224s. The recordings are richly detailed and remarkably spacious.



Sennheiser MD 421 dynamic mics as used by K.G. Attwood in the 1960s to record steam trains. Dynamic mics shine in loud environments.

Ribbon mics are a special type of dynamic mic that use a thin strip of metal foil (the ribbon) suspended between the poles of a powerful magnet to convert sound waves into electrical energy. Generally speaking, they are not used for field recording as they can be easily damaged. They are great for recording voices and musical instruments.

FIELD RECORDING TECHNIQUES: STEREO

Let’s start with stereo recording; while single mic mono recordings play a fundamental role in field recording, most practitioners, especially beginners, will want to capture environments in stereo. Humans have two ears and this binaural hearing allows us to sense location, motion, proximity and elevation. Stereo recordings made with 2 microphones are capable of documenting acoustic content in ways that make 2-speaker – monitor or headphone – listening believably naturalistic. It’s certainly not perfect (there’s a whole chapter later in this book about multichannel surround sound and Ambisonics) but stereo recording/reproduction has been a ubiquitous standard since the mid-1960s. Hence, we are all practised in listening beyond its limitations; environmental sound doesn’t emanate from two points in front of us or from speakers directly over our ears! Different microphones and setups yield results with varying degrees of accuracy, focus, separation and width.



Lucy Wood using LOM basicUcho mics inside Rycote BBG windshields to record the river Thames. (Photo: Marcus Leadley)

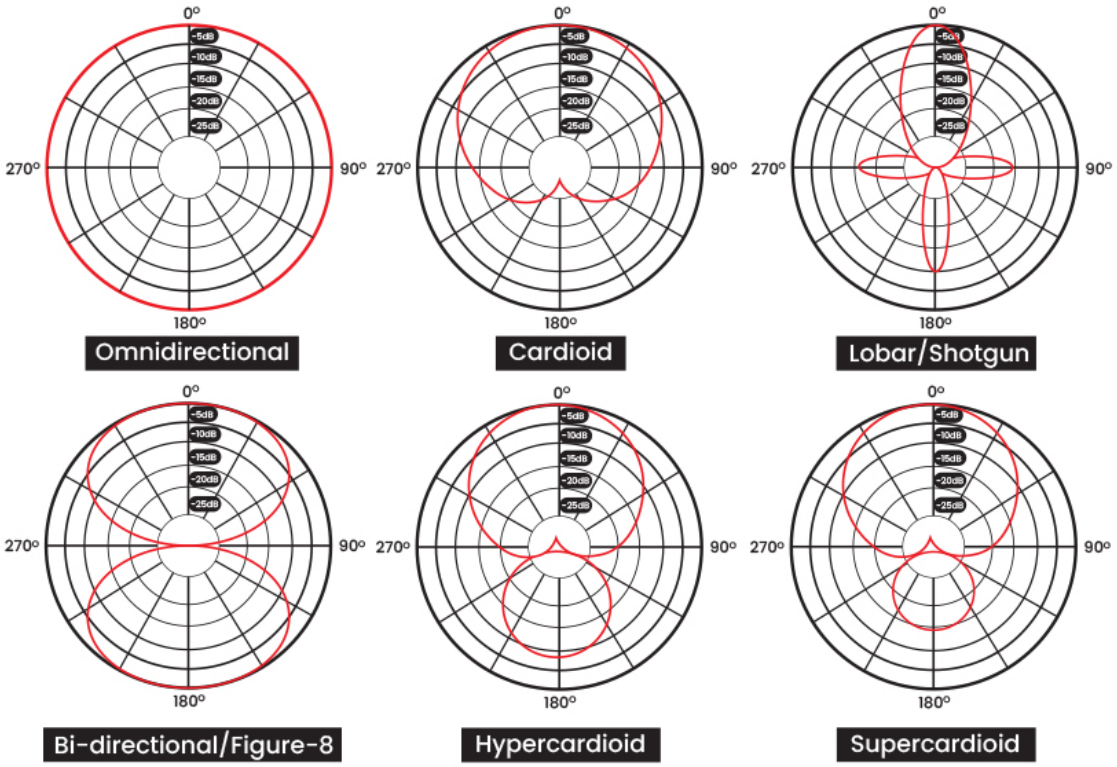
By learning how to manipulate these characteristics, depending on the subject we are recording and the effect we want to achieve, we become phonographers as opposed to people who simply know how to turn on a sound recording device. The majority of the techniques that follow were developed for use in the studio environment. All can, and have, been employed successfully in the field. However, convenience, portability and the minimisation of the 'fiddle factor' are far more of a concern when working on location, so various stereo microphones have been developed – these I'll detail in a separate section. Note that I'll also be addressing wind protection as a separate topic later in this chapter.

While it's neither practical nor necessary to name and compare the gazillions of dynamic and small capsule condenser mics that could be used for field recording (I currently count 30+ brands in the market with prices ranging from £30 to £2000) options exist at all price points. Some of the brands you might like to explore include

Behringer, Superlux, SubZero, MXL, Presonus and T. Bone (low cost); RØDE, Lewitt, Shure, AKG, Beyer Dynamic, Audio Technica and Sennheiser (mid-price); Neumann, Telefunken, DPA and Schoeps (high end).

FACTFILE 2.2: MICROPHONE PICKUP/POLAR PATTERN

Microphones are directional tools, and the polar or pickup pattern indicates how sensitive a mic is to sound coming from different angles or directions. Knowing a microphone's polar characteristics lets you choose the best one for the job and creatively use different microphones to capture subjects and environments in different ways. Polar patterns are documented using a graphical representation: a polar plot. An omnidirectional microphone picks up sound equally from all directions, while a cardioid (named for the heart-shape of the pick-up pattern) microphone is more sensitive to sounds coming from the front and sides, but less so from the rear. Supercardioid and hypercardioid microphones are more directional than standard cardioids while bidirectional (figure-8) microphones pick up sound from two opposite directions: the sides or front and back depending on orientation. So, for example, an omnidirectional microphone (or stereo pair) may be the best choice for recording ambiences and environments. Need good stereo width but want to minimise the sonic impact of a road running nearby? Choose cardioid mics and set them up facing away from the road. If you want to create a focus on a central sound source while limiting the impact of the surrounding environment, choose super or hypercardioid mics.



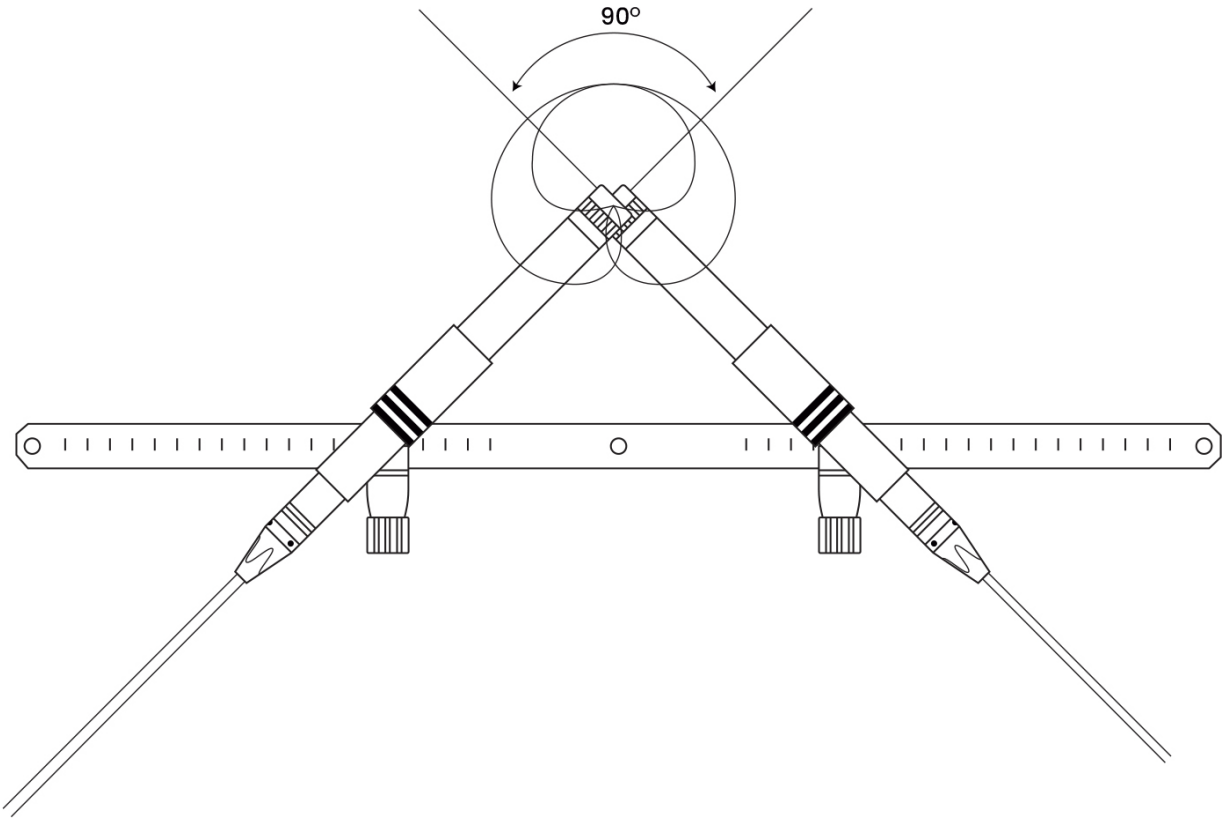
Most common microphone pickup patterns.

Coincident Pair

For most people, a coincident pair (sometimes called an XY stereo array) of cardioid pattern mics will be the first recording option they encounter; the built-in microphones of many contemporary field recording devices we looked at in Chapter 1 are arranged this way. In a coincident pair the mics' capsules are as close together as they can be, crossing at 90 degrees and sometimes adjustable up to 120/130 degrees for a wider stereo field. Putting the capsules this close together avoids phase issues (*see* Factfile 2.3). This also means that the signals from the 2 mics can be summed to deliver a single mono channel. For built-in recorder microphones it's an effective solution which delivers precise imaging and excellent detail without any setup requirements other than recording level management. While it provides a stable directional image, it's slightly weaker at creating an impression of space than a spaced AB pair – which we'll look at shortly.



Shure KSM 141s set up as a coincident pair with windjammers.

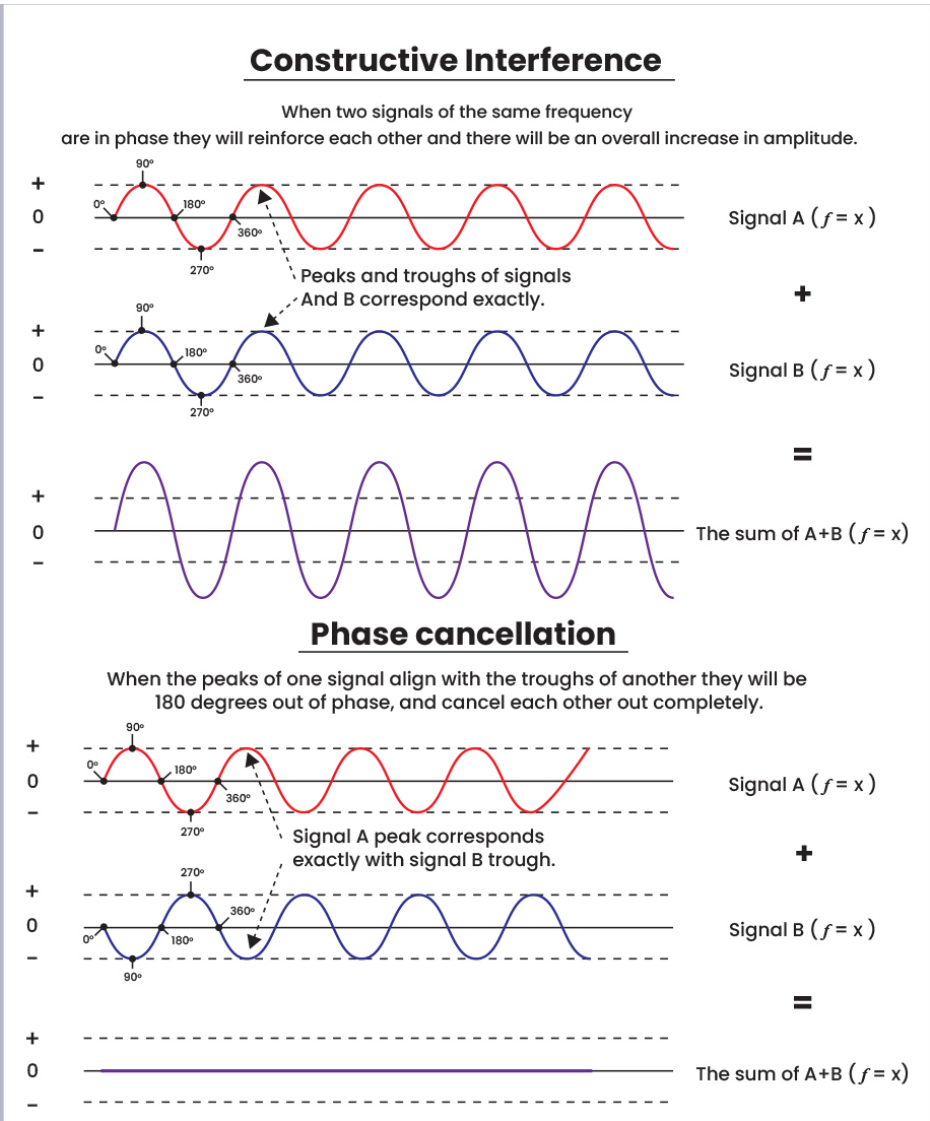


Coincident pair (XY stereo).

With a coincident pair the width of the stereo field is highly dependent on the distance from the mics to the sound source. Stereo width isn't an issue with sounds 2–3m away and general environmental ambiences. However, as we get closer to the microphones, the field width narrows to the point where they act more like a single mono microphone. This can be used to advantage; if you have a voice, instrument or object you want to present front and centre, test the mic position between 30cm and 1m away. Moving them further away will adjust the balance of close and environmental sound. This is a good example of when you want to monitor in the field using headphones. Once you have achieved the desired balance, adjust the recorder's input gain to register between -12 and -6dBFS – unless you are recording 32-bit float (*see* Chapter 1 for more details on level management).

Factfile 2.3: Phase Issues

When you listen to a field recording and you get the sense that ‘something’s not quite right’ with the sound, there’s a good chance you’re experiencing a phase issue. There are two principal causes; direct and reflected sound being picked up by the same microphone or pairs of microphones being too far apart. Managing phase issues in the controlled environment of the recording studio is hard enough – in the field it’s an ongoing challenge.



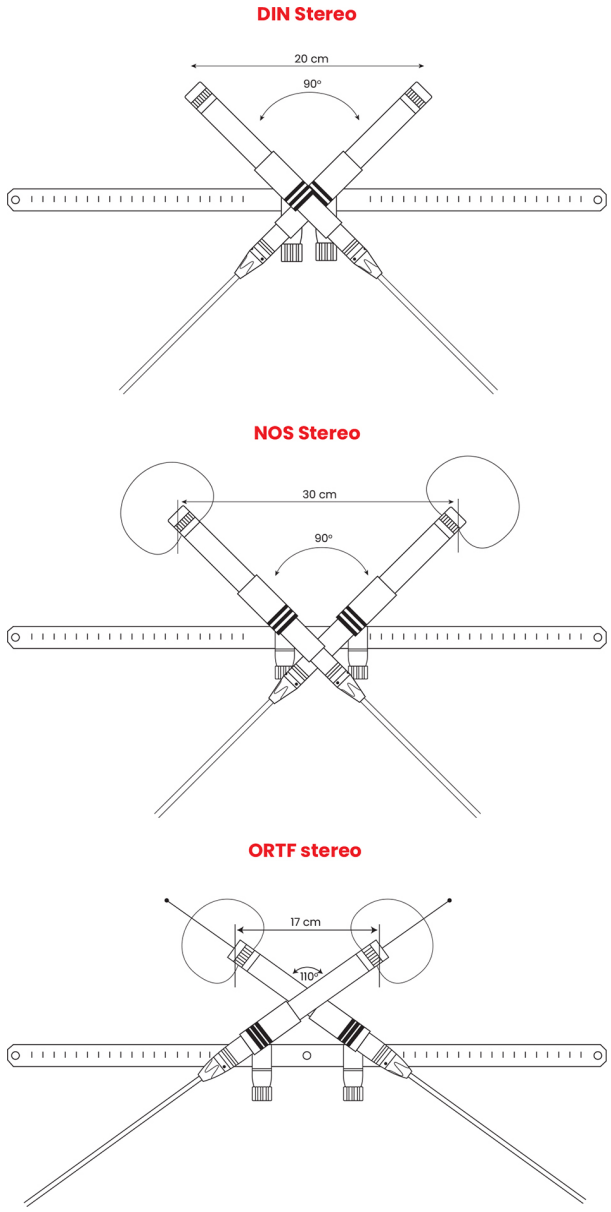
Signal reinforcement and cancellation: phase issues.

The speed of sound in air is not that fast, ranging from roughly 331m/sec at O°C to 344m/sec at 20°C. For humans to perceive two distinct sounds they have to be at least 0.1 seconds apart, so at anything above 17m we start to hear the difference between direct and reflected sound as a discrete echo. With shorter distances we experience it as reverberation (reverb) but do not perceive it as separate from the impulse that created it. When the source/reflection delay is below 20m/sec we experience effects ranging from subtle yet unnatural-sounding tonal colouration to more extreme sweeps of

frequency and issues with sound localisation. These are the result of something called comb filtering: the graphic plot of frequency against amplitude for the summed signals will have so many peaks and troughs that the image is reminiscent of a hair comb.

If we look at the simplest model where two sine waves of the same frequency are being combined, they are ‘in phase’ when the peaks and troughs of the wave coincide exactly. In this situation they will reinforce each other and there will be an overall increase in amplitude. When the peaks of one signal align with troughs of another they will be 180 degrees out of phase, and cancel each other out completely.

Near-Coincident Pairs



Near coincident pairs: three different set-up options.

All these considerations apply whether you are using built-in mics or independent cardioid-pattern mics. Unless you are using very low-cost external mics, they should deliver a higher-fidelity recording than built-in mics. A matched pair is required, and most

manufacturers sell these for stereo recording. It's critical that the mics are set up correctly; even small errors can impact the recording. To do this you need to mount the mics on a stereo bar and ensure that the capsules are aligned, and that the angle is set properly. Stereo bars come in various sizes – short or long, with either fixed or movable mounting positions. For creating coincident pairs, a short bar (20cm) with fixed mounts will be the most convenient. An excellent alternative to the traditional stereo bar is Shure's A27M Stereo mic mount (around £70); as well as creating coincident pairs it can be adjusted to deliver five different near-coincident and near-spaced stereo arrays.

When microphones are spaced less than 30cm apart, the setup is generally referred to as a near-coincident pair. Again, this technique calls for a matched pair of cardioid mics and it's useful when you're trying to capture a sense of the wider environment while needing to maintain a strong central focus. For example, a babbling brook, feeding time at the zoo or a conversation on a beach. There are several popular standards, named after the broadcast organisations where they were developed:

DIN (Deutsches Institut für Normung) technique, which spaces the cardioid-pattern microphones 20cm apart, at an angle of 90 degrees to each other.

ORTF (Office de Radiodiffusion-Télévision Française) uses a pair of microphones, spaced 17cm apart and angled 110 degrees to each other. This is well suited for reproducing stereo sound; the spacing of the microphones approximates the distance between the human ears, and the interaction of the cardioid polar patterns creates a 'shadow' at the centre of the stereo field that emulates the effect of the human head. So, when reproduced on speakers or in headphones, the recording sounds true to life.



Near coincident pair: NOS stereo.

NOS (Nederlandse Omroep Stichting) utilises a pair of condenser microphones spaced 30cm apart and angled 90 degrees. This technique takes us away from the human-centred approach to the mic spacing seen with ORTF – however the tighter angle helps to constrain the spread of the stereo field. Here, the difference between inter-stereo time and level information is more apparent so, subjectively, the recording will sound more expansive, with a pronounced sense of movement.

RAI (Radiotelevisione italiana) is similar to ORTF; here the microphones are placed 21cm apart and angled at 100 degrees.



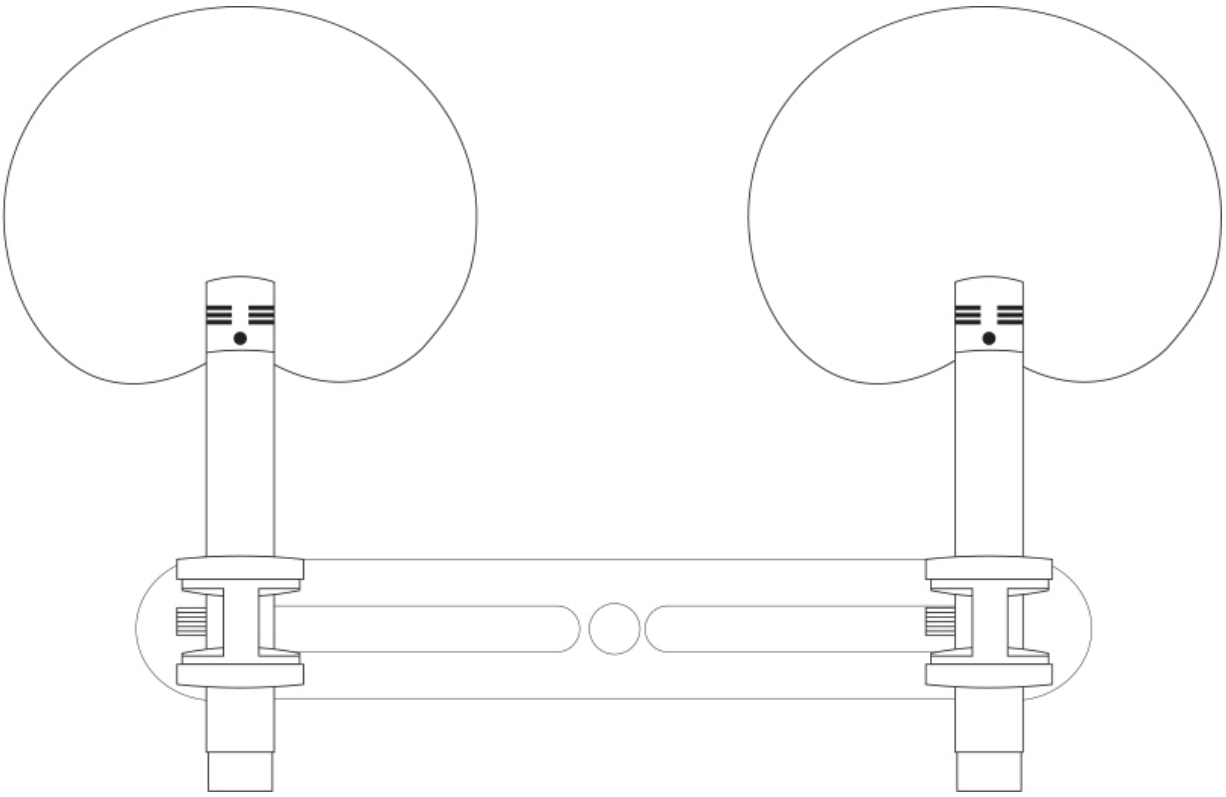
Near coincident pair: ORFT stereo.

Olson stereo has the microphones separated by 20cm and angled 135 degrees to each other. This is the widest angle of any of the formal techniques and this can lead to a lack of clarity/focus at the centre of the stereo image.

Spaced Pairs (A/B Microphone Technique)

When mics are more than 30cm apart the setup is referred to as a spaced pair, and they are useful for recording more diffuse ambiences. As there is no central sound source the mics are set up in parallel. A stereo bar can be used or independent stands if greater inter-mic distances are required. Different spacing will impart different characteristics to your recording and many factors come into play when deciding how far apart the mics can be; what you're recording, the way sound sources are dispersed in the environment, general environmental characteristics such as topography and the presence of reflective surfaces and the speed of sound itself. As

soon as you separate microphones, there will be some time delay of the sound from a single source arriving at each capsule and this can cause phase issues (Factfile 2.3). Another issue may be an apparent ‘hole’ in the centre of an otherwise wide stereo field. With cardioid mics, a progressive degree of frequency roll-off for off-axis sounds is accompanied by some tonal colouration: essentially sounds on the periphery will be quieter and less bright. Different types of mics will give different results so spend some time getting to know the ones you have: experiment with distances between 1–3m so as to become familiar with the difference in stereo width, the tonal colouration and the effects of time delay.



Spaced cardioid mics on stereo bar.

Spaced pairs deliver wide stereo images but cannot be summed to mono effectively (you can of course use just one side of the pair as a mono source). In broad terms the sound image will be less precise than with a coincident or near-coincident pair. Details can be

a little more blurred, giving more of an overall impression rather than true accuracy. However, depending on the effect/feel you want to create, this may be exactly what's required.

Spaced Cardioids

With cardioids there is a general rule for studio deployment that two identical microphones should be separated by one-third to one-half the width of a sound stage and pointed directly at the primary sound source. This translates directly to field recording situations where there is some form of focal point – such as a group of animals in a field – but not for recording more expansive soundscapes/environments. For these, XY, A/B stereo or spaced omnidirectional mics are more appropriate. Using spaced pairs in the field requires a degree of fine-tuning by ear because every location is unique. Again, this makes headphone monitoring essential.



Setting up a pair of AKG C451s as a spaced pair for indoor recording – a stereo bar will be more practical in the field.

If you're recording in an enclosed space such as a factory, glasshouse or tunnel, the microphones should be located equidistant from fixed boundaries (such as walls) to avoid differences in any reflected sound (reverberation or echoes) negatively impacting the recording. Having an assistant that can move microphones while you monitor is extremely useful. Start by 'sound checking' one mono mic to identify the ideal location for your desired result; a mic in the middle of a space will pick up a very different version of the acoustic environment to one in a corner. In some situations, placing the microphones very close to the wall or floor will be advantageous because of the boundary layer effect (*see* Factfile 2.4). In outdoor settings we also need to pay attention to reverberations from cliffs, buildings or other environmental factors. In a situation where there are more pronounced echoes, the mics should be deployed at the same distance to the reflective surface, but spaced along a horizontal axis. Whatever the final setup, remember to aim the microphones at the 'action'.

Spaced Omnidirectional Mics

The use of spaced pairs of small or micro omnidirectional condensers is one of the most exciting areas of field recording development. For many years omnidirectional DPA 4060 micro condensers have been a ubiquitous feature of the professional kit bag. They are excellent sounding mics – as pointed out earlier in this chapter, smaller low-mass diaphragms are the fastest in terms of response times. As a result, a pair of 4060s can capture natural environments in stunning detail. Much of Chris Watson's work for David Attenborough's BBC series has been achieved using 4060s. He has a wonderfully simple trick of attaching a pair to a coat hanger and hanging it in a tree: brilliant results and ultimate portability. The 4060s are also extremely versatile; being about half the size of the rubber you find on the end of a pencil means they can literally go where other microphones cannot and are easy to hide from view if

required. They can be used as lavalier (lapel) mics and ship with a useful selection of surface mounts.

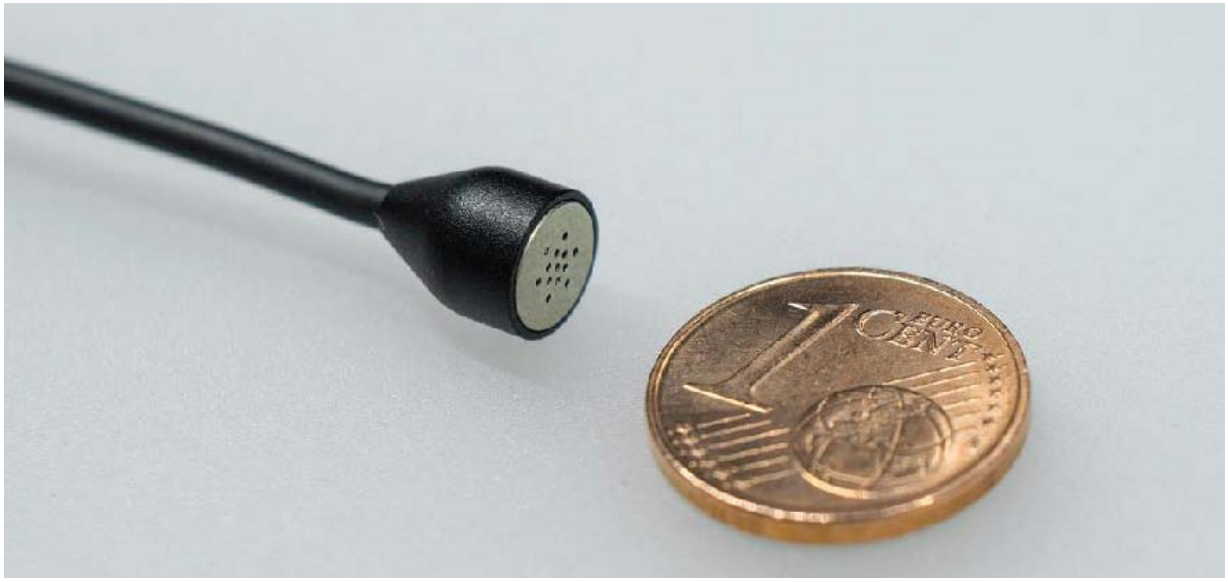


DPA 4060 kit. Mics come with various mounting/attachment options that cover most situations.



Pair of omnidirectional DPA 4060s mics attached to coat hanger and hung from a tree: a Chris Watson technique that’s easy to deploy and delivers great results.

One of the main reasons why the use of 4060s hasn't filtered down to the general field recording community is price (£900+ for a pair), especially when they are so tiny and look so fragile. However, I've always found them to be surprisingly robust. If you are really planning to give your DPAs some serious grief there is also the 4660 series heavy-duty omnidirectional microphone.



LOM mikroUši. Small mics like these are useful as they can go where larger mics cannot fit.



LOM basicUcho. The mic is built in to an XLR plug enclosure for convenience.

Two companies, LOM and FEL MicBooster, manufacture small omnidirectional microphones, with pairs costing in the region of £130 in different formats: with enclosures and cables – LOM Uši and Uši Pro and MicBooster's Clippy XLR EM272Z1; and there are also versions built directly into XLR plug enclosures – LOM basicUcho and MicBooster Pluggy XLR Primo EM272Z1.



MicBooster Pluggy EM272Z1 on stereo bar for recording riverside ambience.

Both LOM and MicBooster units feature the highly regarded Primo EM272Z1 electret capsule. And if you fancy a DIY project, MicBooster also sell various capsules and enclosures separately. All

these mics offer excellent sound quality and the benefits of small size in a low-cost package. I certainly won’t be getting rid of my DPAs any time soon, but I’m very impressed with both LOM and MicBooster products and use them in all sorts of situations.

Deepal Warakagoda has been a field sound recordist for over two decades, originally focusing on birds and other animals on the tropical Indian Ocean island of Sri Lanka. His recordings have appeared in several publications, both local and international, and in global sound archives. His interest later moved to capturing sounds of fauna in stereophony to represent their presence in nature in a manner even closer to reality. Recently he initiated a sound map project of Sri Lanka to present stereophonic soundscapes of its natural habitats as an audio tool to aid future nature conservation programmes. He is a contributor to the *Sounds of Your Park* global project.



Deepal Warakagoda recording birds and other animals in Sri Lanka using LOM microphones. (Photo: Akila Abeyratne)

LOM supports Deepal's project via its LOM+you programme. For more information see

www.srilankanaturesounds.com

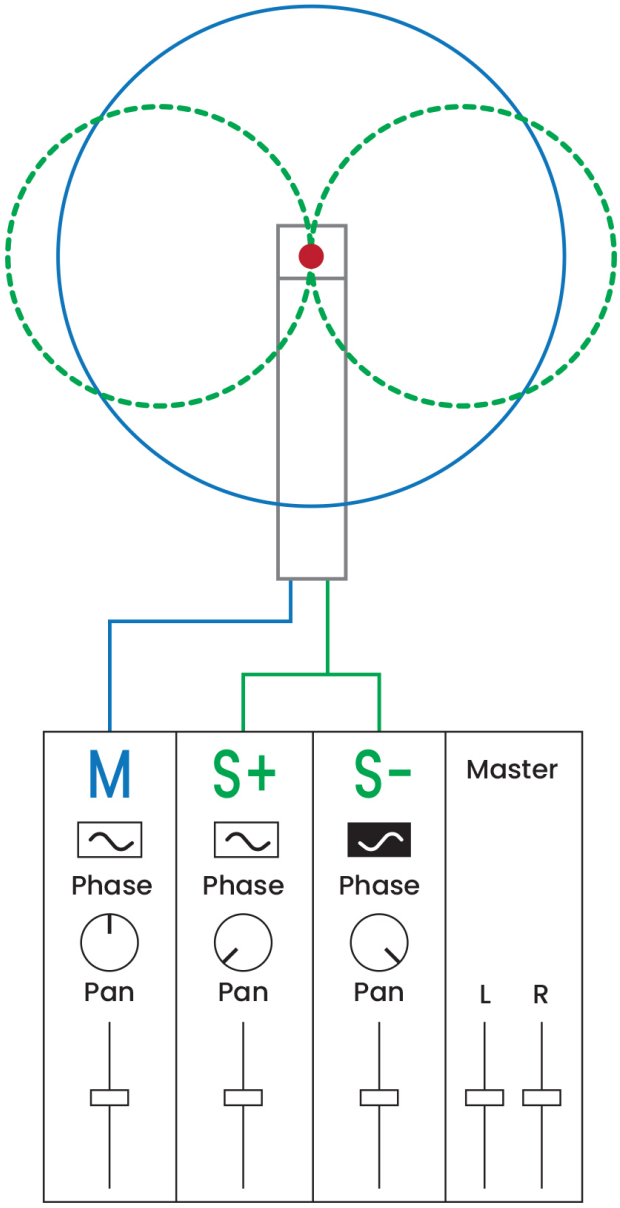
Baffled Spaced Pair

If a baffle made from an acoustically absorbent and non-reflective material is placed between microphones in a stereo setup, the channel separation will be enhanced because off-axis sound for each mic will be attenuated. This works with all of the stereo techniques so far discussed (A/B stereo, ORTF, DIN, NOS) that use cardioid pattern mics and also with omnidirectional mics. It is most effective when the mics are fairly close together; to some extent the baffle mirrors the role of the human head in binaural hearing. One specific baffling technique uses something called a Jecklin Disc – named after the Swiss sound engineer Jürg Jecklin who invented it. Here, two omnidirectional microphones are spaced 17.5cm apart and a disk-shaped baffle 35cm in diameter is placed between them.

Mid/Side Stereo

Mid/side recording is an extremely useful technique that delivers material that can be used for mono sound or variable-width stereo. As with most of the techniques we've looked at it was developed for studio use (by Alan Blumlein) but is easily adapted for use in the field. Two different microphones are required (or different capsules in the same mic – as is the case with an M/S shotgun mic we'll look at in the next section of this chapter). Firstly, a mid mic, which is most commonly a cardioid – you can however use a hypercardioid or omni mic depending on the stereo image you want to create. Using a cardioid-pattern mic will yield an outcome comparable with an XY pair. This mic needs to be set up pointing directly at the subject or principal focus of the recording. Alternatively; using an omni mic will deliver a more overall sense of environmental ambience with less of an obvious central focus. Secondly, the sides-mic must have a figure-8 polar pattern; this needs to be set up at exactly 90 degrees to the mid mic and both capsules need to be as close together as possible so as to avoid phase issues, and vertically aligned. In the

field this can be problematic due to the need for wind protection, but there are special blimps/mounts available. And there are clear benefits to using a mid/side shotgun in a blimp mount in the field.



M/S stereo: two mics, or a single mic with two capsules, can create a mono image or variable-width stereo field.

However, if using individual mics, position the sides mic with its ‘front’ lobe pointing to the left with the null axis aligned with the mid mic so it does not pick up what the mid mic captures. To monitor M/S

stereo in the field the signal has to pass through an appropriate decoding matrix, otherwise all you hear is the mono (mid mic) on one channel and the figure-8 on the other. Thankfully some (but not all – so check) contemporary recorders such as the Zoom H series units and Sound Devices Mix Pres offer an M/S monitoring option for stereo sound. On a practical point, don't expect the mid and side signals to be at a similar level; the side signal will be quieter and there is no need to try to gain match the signals on your meters.



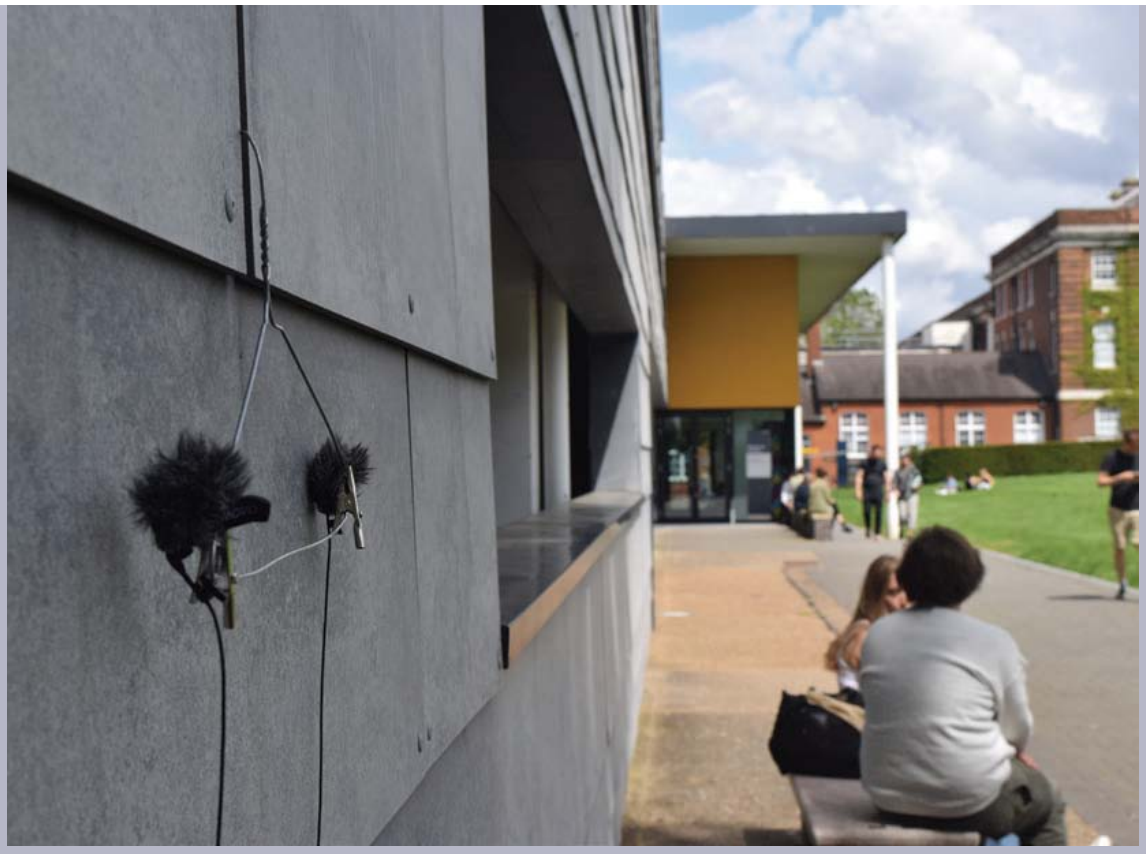
Sennheiser MKH 418-S M/S stereo shotgun in blimp mount – wind cover removed to show detail.

So, what is happening here? M/S stereo works because each side of the figure-8 mic captures sound, but the polarity is reversed one to the other. The easiest way to explain how this is used to create a stereo image is to look at the process traditionally used in the studio for handling M/S stereo recordings. Nowadays, there are plenty of encoder/decoder plug-ins to do this but before this, working with M/S stereo required you to set the mid mic recording on one

channel panned centrally and then duplicate the figure-8 recording. These were then placed either side of the mid channel, panned hard left and right and the phase of one side was flipped (many analogue desks had a button for this) to give the positive and negative signals required. Then, when the side channels were lifted, the recording would in an almost magical way develop a stereo characteristic achieved as a result of phase cancellation between the mid and side signals.

Factfile 2.4: Boundary Layer Mic'ing

Physics is such that sound pressure doubles along a sound-reflecting surface. Placing a microphone in this boundary layer enables it to take advantage of a natural boost in sensitivity of 6dB for direct sound. Also, and perhaps more importantly, there is the potential to eliminate comb-filtering effects: this is the tonal colouration created when direct and reflected sound reaches a microphone at the same time. A mic in free space or close to a wall or floor (but not in the boundary layer) will receive both direct sound from a source and its reflection as it bounces off the surface and back to the mic. The extra distance travelled by the reflected sound will create a delay – longer or shorter depending on the distance travelled. Placing the microphone flush with the boundary ensures that those reflections can no longer interfere with the direct sound, and hence the recording sounds more natural.



DPA 4060 located by a wall to take advantage of natural boost in sensitivity created by the boundary layer effect.



AKG PZM boundary layer mic (circular) installed above studio control room window to capture room sound.

However, this boundary layer is quite narrow so you cannot simply put a Zoom recorder near a wall and hope to benefit – although, minimising reflection delay may improve sound quality. Small omnidirectional condenser mics such as DPA 4060 and electrets by LOM and MicBooster are useful for this – and even Soundman OKM suggests a secondary use for its binaural mics as boundary layer mics. You can also buy dedicated boundary layer mics – often referred to as PZMs (Pressure Zone Mics). Most purpose-built PZMs are contained in a low-profile housing (such as Audio Technica Pro 44, Shure Beta 91 and Sennheiser e901) or attached to a thin, flat plate (Crown PZM 30D or AKG PZM6 D) for easy installation. PZM mics are not specifically designed for field recording purposes – they are often installed into lecture theatres and music practice rooms as an easy solution for capturing the full room sound with a minimum of setup and to take best advantage of the boundary layer effect. However, boundary layer mic'ing can be

very useful when recording in larger indoor spaces or in environments where a specific surface (a cliff, building or other structure) is an obvious source of problematic reflections.

Looking further into the physics, boundary layer mic'ing offers additional benefits. While sound pressure for direct sound gains the full 6dB increase, reflected sound only increases by 3dB at the boundary due to phase relationships in the diffuse sound field. Hence, recordings will seem marginally less reverberant than recordings made using freestanding omnidirectional mics. Finally, a word about the role of the boundary itself. The 6dB gain will not be consistent across all frequencies unless the dimensions of the boundary are roughly the same size, or greater than, the wavelength of the low-frequency content you want to capture. For the most part, using walls or floors achieves this with ease.

Stereo Microphones

Stereo microphones offer a number of benefits for working in the field; they are quick to set up and there are no issues with capsule alignment as this is already designed in. The only real disadvantage is that you will have to pack additional independent mics if they are needed for mono or greater stereo separation.

For those looking for a budget external XY mic, Superlux has the E523/D. With a frequency response of 30Hz–20kHz it punches well above its £60 price tag. If you're just starting out, but happen to be using a recorder without built-in mics, it could be a really useful addition to your kit bag. However, the built-in mics of a Zoom or Tascam unit will deliver a similar result.



RØDE NT4 in blimp suspension mount with wind protection removed. The coincident mic capsules will record an excellent stereo image without any phase issues.



Audio Technica BP4025.

At around the £450 mark, the RØDE NT4 is an excellent-sounding and robust option. The small cardioid condenser capsules are arranged as a 90-degree coincident pair and the mic can be powered by phantom or a 9V battery. I've done a lot of work with one of these and for me, the only real downside is weight; holding a 480-gram mic in a blimp mount can become quite tiring by the end of a long day. Another mic with a coincident pair design (this time mounted inside a larger single enclosure) at around the same price

is Audio Technica's BP4025. The company specifically designates it as a field recording mic and while its top-end frequency response isn't quite as good at the NT4 (17kHz as opposed to 20kHz), this shouldn't be an issue unless you are planning to do some radical downward pitch shifting. And you are looking at a very low self-noise floor and a weight of only 247 grams.

There is also the Audio Technica AT8022. This was originally launched as a lower-cost stereo mic at nearly 50 per cent of the cost of the BP4035. However, prices in the market have drifted closer together to a point where there is often less than £80 difference in cost. The AT8022 is a very capable mic that can be battery powered if you don't have a recorder/camera capable of delivering phantom power. At 15kHz, the upper limit of frequency response feels perhaps just a little on the low side. Another option at around the £1000 mark is the Shure VP-88 – this time a mid/side stereo mic (*see* the section on M/S stereo earlier in this chapter for a detailed explanation). The VP-88 contains an internal M-S matrix with three settings for different degrees of stereo image separation. The internal matrix can be bypassed if stereo imaging is to be done in post, and it can operate on phantom power or an internal 6V battery.



Schoeps MSTC 7 ORTF.

If you're interested in exploring ORTF stereo in the field, the Superlux S502MkII is an ideal beginners' solution – and the

performance is such that it’s a capable stereo mic in many field recording situations where a price tag of around £130 makes it the ‘riskable’ option. Whereas at around £3000, the Schoeps MSTC 7 ORTF (shipped with Mk4 capsules or without if you already have a matched pair of Schoeps cardioid capsules) is unlikely to be the first mic you’d consider leaving on the riverbank overnight. Sonically of course, it’s in a completely different league.

Stereo shotgun microphones are also extremely useful in the field. In terms of enhanced directionality, they rely on exactly the same design principles as the mono shotgun microphones I discuss later in this chapter. However, rather than contain a single mic capsule, two are employed to create a stereo image. Contemporary stereo shotguns are designed for mid/side recording. These have become extremely popular in the film and television industry because a single microphone’s output can provide mono and variable width stereo audio, which gives excellent flexibility in postproduction without the need to deploy multiple mics in the field.



SSH-6 mid/side stereo shotgun capsule.



Zoom SSH-6e stereo shotgun capsule.

Until recently there was no low-cost option, with stereo shotguns starting at around £900. Thankfully Zoom filled this gap in the market with the release of the SSH-6 (around £125) capsule, which is compatible with all Zoom's H5, H6, Q8, U-44 (interface), F4 and F8 recorders. There is also the SSH-6e version for the new Zoom 32-bit recorders. Like other contemporary devices, they are both mid/side stereo shotguns.



Audio Technica BP4027.



Sennheiser MKH 418-S.



Sanken CSS-50.

Up in the £900+ bracket we find Audio Technica BP4027 and its slightly shorter sibling the BP4029 (236mm as opposed to 380mm – the longer the tube the more directional the microphone) which both offer three selectable settings: left-right stereo output (wide or narrow) via the microphone's internal matrixing system or as discrete mid/side signals. This is extremely useful; sometimes you just want a L–R stereo recording. Sennheiser's MKH 418-S (around £1400) is something of an industry standard, ideal for field recording and all applications relating to film, radio, television and so on – no internal M/S matrixing however, so purely an M/S shotgun. The Sanken CSS-50 is another high-end professional solution. At 220mm it's a comparatively short shotgun (which is convenient and practical) and

the unique 3-capsule design allows for three different settings: Mono, for sharp shotgun directivity and excellent low-frequency definition; Normal, for precise directional hypercardioid focus with accurate stereo localisation and Wide for an expanded 140-degree stereo image.

Binaural Stereo

Binaural recording is a technique designed to take full advantage of the way we perceive sound – as such it can create a realistic sense of space and motion, even an apparent sense of events taking place in front, behind, above or below the listener. Many people's first experience of binaural stereo is surprisingly impactful; we are simply not used to hearing recordings that sound truly 'real'. There is a catch, however – it only works with headphones, the complex data that's encoded doesn't translate to monitors. You can listen to binaural recordings over speakers, but you won't get the full effect and the stereo image may feel unnatural. If you reference back to the section on human hearing at the start of this chapter, you'll see how binaural recording takes full advantage of the human physiology. Binaural mics, small omnidirectional condensers, are worn in the ears. This generates the necessary 'head shadow', creating a fractional delay between one ear and the other. It also causes tonal colouration and differences in amplitude that the brain decodes as directional information. These factors are further enhanced by the shape of the pinnae, enabling the recordings to become highly nuanced – providing the listener with a natural sense of distance, direction, position and orientation.



Binaural recording – Roland CS10-EM in-ear mics and SoundMan windshields. Wind protection is generally needed even on the calmest day, especially in exposed locations.

For the most accurate listening experience, the listener should be the person who made the recordings. In practice, the mechanisms of human hearing are such that we can still decode cues processed through other head-sizes and pinnae dimensions. However, there are subtle, discernible differences and the idea of listening ‘with other ears’ is something I find truly fascinating. As well as sound quality, binaural recording is a very practical and portable option. Also, people will assume you’re simply listening to music on earbuds – so you are practically invisible. This can be appealing in many situations but the idea of ‘stealth’ recording raises a number of ethical considerations. For more on this issue, *see* Chapter 5.

Several companies make microphones specifically for binaural recording. Many of the solutions are surprisingly low cost, which is excellent news for anyone on a budget. There are basically two types; those that are just microphones (Soundman OKM, Sound Professionals and DPA) so you can’t monitor sound while recording

(you can still hear what is happening around you of course) and those with built-in monitors (Roland) – both have advantages and disadvantages. With those without monitoring it's really easy to forget you are actually recording and this can lead to very natural-sounding results. On the other hand, those with built-in monitoring make you acutely aware that you are recording. As a result, you tend to be led by the action – moving towards sounds and acoustic spaces that may appear more interesting. Hence the recordings have quite a different character, and can become more of a 'performance'.



Soundman OKMII kit with A3 preamp. The preamp is only needed if you're using a recorder that doesn't support plug-in power.

OKM Soundman has been making binaural mics since 1985 and with prices ranging between £60–£180 remain an excellent solution. Both the company's OKM1 and OKMII are suitable for field recording, with the II having the higher specification. All capsules are matched pairs. For particularly critical recordings, the Studio series receives further inspection and ships with a test certificate showing the differences between left and right-side performance. There is

also a Rock Studio version designed to record high SPL locations such as rock concerts, with a max SPL of 142dB which is useful for recording industrial soundscapes and motor racing.

Soundman also make various accessories. There's the A3 preamp that can provide plug-in power for the electret capsules. You can buy this separately or as part of a kit with the mics. Technically, it's not required if you have a Zoom H-series recorder or some newer-generation Tascam units as these have a 3.5mm socket that supports plug-in powered devices. However, the A3 allows you to connect the mics via a recorder's combo inputs (with the appropriate break-out cable) and has a built-in bass roll-off switch and an attenuator (0 to -20dB) to limit distortion when recording high-level signals. It has its own battery so it reduces draw on the recorder. Alternatively, there is the Soundman PPA unit that allows the mics to be powered via phantom. And there is also an adaptor to allow the mics to be used with an iPhone or Android device.



Core Sound binaural mic set. Comes with its own preamp.

Soundman make a binaural windshield (around £100) that looks like a pair of fluffy earmuffs – but they totally do the job; in-ear mics are extremely prone to wind noise. On a final note, it's worth adding that the OKM mics (and most binaurals) can be used as boundary layer mics or as livelier mics, as required. Sound Professionals have the SP-TFB-2, low noise in-ear binaural microphones, offering a choice of sensitivity and connector type for between $100–200. I'm not aware of any European distributors so one must order direct from the US. Core Sound has a binaural microphone set (from $230) which includes its own dedicated preamp. You can specify the mic response (fixed flat or fixed bass roll-off filter), plug and clip type.



DPA 4560 CORE Binaural Headset. Easy to wear and delivers excellent DPA quality sound.

If you're looking for top-end quality the DPA 4560 CORE Binaural Headset really is a fantastic solution. A pair of 4060 miniature omnidirectional microphones comes mounted into the two 'earhooks' of a type 4266 Flex headset. The price is around £900 and you will also have to buy a pair of microdot to XLR adaptors.

For mics with a built-in monitoring option, Roland's binaural CS10-EM are very cost effective at around £80. Again, the recording quality is very good. Care needs to be taken in setting levels as feedback between mic and speaker can easily happen. Unfortunately, the very useful Sennheiser Ambeo Smart Headset

that worked for binaural recording with iPhones was discontinued in 2024.



Roland’s binaural CS10-EM. These are equipped with in-ear monitors so you can hear what you are recording at the same time.

There’s also an alternative option for binaural recording that uses a dummy head – very useful for durational/stationary recordings. These heads (and other frameworks) often have built-in microphones. This form of binaural recording can attract a lot of attention, so definitely not a stealth option! AWI offers two versions of its dummy head, a standard SR3D Mk III model (£320) and one with enhanced weatherproofing (£390) designed for long-term installation. Both are supplied with Primo EM272 electret microphones and XLR, 3.5mm or USB connectivity options. Soundman also offers a dummy head – this comes without mics and is priced between £500–£600. Neumann’s KU100 (£7,500) comes with built-in mics, two-stage switchable subsonic filter, switchable

10dB pre-attenuation and balanced and unbalanced outputs on both XLR and BNC connectors.



Soundman dummy head for binaural recording. An alternative to in-ear mics. Ideal for fixed location recording.



3DIO Free Space Binaural Microphone (also available in all black). A ‘dummy head’ doesn’t have to look like a head in order to record binaural sound.

The AWI, Soundman and Neumann options closely mirror the appearance of the human head, but this is not essential for binaural recording. More important is the inter-ear distance and the masking effect or head shadow. AWI offers a range of biannual cubes and tubes with fixed ‘ear’ spacing and a version where you can adjust the separation. AWI heads and tubes are also available as DIY kits at a significantly reduced price. Two options with built-in microphones are offered by 3Dio; the Free Space Binaural Microphone with electret capsules and the Free Space Pro II, which is fitted with DPA 4060 omni condenser microphones.

Dallas Simpson is an environmental binaural performance artist who has been using in-ear binaural microphones since the early 1990s. This was a technology he came to after many years of experimentation. In terms of locations, he doesn't have a 'recipe environment' but looks for places with different elements, maybe tunnels, bridges, rock faces, cliffs or a stony, rocky shoreline with sand in places. This enables him to move between different kinds of environments and explore the sonics. From the point of view of improvisation, he looks for ways of connecting physically with these locations to elicit sounds. He interacts with objects and surfaces but doesn't 'play' the environment, preferring to 'sound' the location, honouring and respecting its physicality. Sometimes he will also prepare an environment in the way John Cage might prepare a piano. He uses titanium sheets as acoustic sound mirrors to spatially choreograph a soundscape, allowing sounds from lower down (such as a stream) to blend with sounds higher up (such as wind in the trees) – this is an approach that can be heard on his *World Listening Day 2020 – Lambley Dumbles* release (https://impulsivehabitat.com/releases/ihab143.htm). He also uses acoustic listening tubes to bring sounds directly to his ears/mics so as to interrogate surfaces at close quarters.



Dallas Simpson, installation, performance and recording near Brick Lane, London (2005). Note binaural in-ear microphones.

MONO RECORDING

It’s easy to think of mono recordings as somehow inferior to stereo: something of a throwback to the 1950s or earlier. However, this is very much not the case. For any subject where a strong central focus is required (such as the human voice), mono recording is a go-to solution. Mono recordings are often the most useful when creating sound effects and ideal for the focused documentation of birdsong and other biophonic sources. Mono recordings can also be placed anywhere in the stereo field of a lay-up or composition and they are comparatively easy to automate in software to create pseudo-realistic motion events. And they can easily be placed in artificial spaces using digital reverbs. Shotgun mics are a type of highly directional mono microphone (I’ve already talked about stereo shotguns in some detail) often used by field recordists.

Mono Shotgun Mics

Harry Olson, an acoustical engineer at RCA Victor, developed shotgun mics for use in the film industry in the 1950s. When you see a film set and someone has a mic on a boom pole over the talent, it will almost certainly be a shotgun. They’re still used for this purpose but have largely been replaced/augmented with wireless lavalier (lapel) mics that can be discretely hidden in clothing or taped to the skin. Shotgun mics are designed to pick up sound from a specific direction while minimising sounds coming from other directions. The name comes from their long, cylindrical shape – sometimes they’re referred to as rifle mics. In a traditional-design shotgun microphone the capsule itself, usually a supercardioid condenser, is mounted inside the body of the mic at the base of what’s called an interference tube. Look closely and you’ll see that there are slits cut into either side of this tube, and these groups are out of alignment with each other. On-axis sound directly in front of the microphone

(the sound you want) travels straight down the tube to the capsule's diaphragm. Off-axis sound (the sound you don't want) has to enter the mic via the side slits, and the different distances the sound has to travel creates varying phase relationships, resulting in partial cancellation of sound.



Mono shotgun mic in blimp recording seabirds. Directional mics remain an important part of a field recording kit.

It's a clever mechanical design but by no means a perfect solution; the amount of cancellation is frequency dependent – reasonably efficient with high frequencies but only very long, specialist shotgun mics are directional at low frequencies. Also, any side-sound that is captured can have significant tonal colouration. Moving sound, or moving the mic can lead to phasey-sounding recordings. So shotgun mics are best suited to situations where the

off-axis sound is significantly different to the on-axis sound and the mic can be mounted in a fixed position or held relatively still.

Shotgun mics come in various different lengths, with short shotguns gaining popularity due to convenience and this is reflected in the wide range of options on the market. Unfortunately, physics is such that longer interference tubes are simply more efficient, so short shotguns are only good at rejecting side-signal upper mid- to high-frequency content. There are exceptions – but these come at a price. For example, the Schoeps SuperCMIT 5 has much stronger directionality and better broadband frequency rejection than one might expect from a microphone of its length because of its twin capsule design and on-board digital signal processing capability. However, with a price tag exceeding £3200 it's a lot more expensive than a very capable RØDE NTG3, Sennheiser MKH416 or Shure VP89 – all of which use RF-bias technology (as sound waves move the diaphragm, the change in capacitance modulates the tuning of an RF [radio frequency] oscillator, which is then converted back to an audio signal by an RF demodulator circuit) which makes them far less sensitive to humidity and moisture.



Schoeps SuperCMIT 5.



RØDE NTG3. RF-bias technology makes mics like this far less sensitive to humidity and moisture.



RØDE NTG1. A great all-purpose shotgun at a good price.



Sennheiser MK 600. Marketed as a camera mic but very popular with the general field recording community.



DPA 2017.



DPA 4017.



Audio Technica AT897.

There are plenty of good-quality standard-design short condenser shotguns in the £200–£300 price bracket, such as the Shure VP82, RØDE NTG1, Sennheiser MKE 600 and the Audio Technica AT897. For Zoom recorders, which take interchangeable mic capsules, there is the SGH-6 for around £100. There are also plenty of options at higher price points, such as the Neumann KRM 81 i (for medium distance) and the KRM 81 i mt (longer distance). DPAs 2017 is an extremely directional mic built for durability and consistent performance in cold weather and humid environments. The company's top-of-the-range 4017 can be supplied with one of four different preamps, depending on application.

In practical terms, what does all this mean in the field? Well, it suggests that if you're trying to record the sound of, say, a single cricket in mono in an environment filled with crickets, you may be better off choosing a conventional supercardioid mic (such as a Beyer Dynamic MC950, Rycote SC-08 or Schoeps CCM41) rather than a mono shotgun; any benefit of enhanced side rejection is likely to be minimal, plus there is the danger of some tonal colouration from any off-axis sound that is captured because of the interference tube. Similarly, using a short shotgun to record machine sounds in a reverberant factory space or one filled with low-frequency noise isn't the best solution. It's better to get a super/hyper cardioid condenser as close as possible to the sound you want to capture.

If the sound source is loud enough, consider using a dynamic microphone to better exclude distant sounds and ambiences. Recording pristine rainforest ambience where you want to capture the whole environment? Not a job for a shotgun – a spaced pair of omnidirectional mics is likely to be the solution. On the other hand, if you are looking to capture the sound of a specific bird or animal in a tree from a distance, tools in use on a workbench in a quiet shed or a person speaking in a field, a shotgun is a great solution. Now we begin to see how empowering knowledge of microphone characteristics can be.

Parabolic Microphones

For situations where an even greater degree of directionality is required than can be provided by a shotgun mic, especially for distant sounds, there is the parabolic microphone. The microphone itself isn't parabolic; the mic capsule, usually a cardioid or omnidirectional condenser, is placed at the point where a parabolic dish reflector focuses sound. The mic points into the dish, not in the direction of the sound itself. The dish acts as a mechanical amplifier (up to roughly six times) and peripheral sounds are either not amplified or blocked by the dish itself. They are very useful for recording birdsong and in sports-sound applications. Parabolic mics

also have a role in surveillance and detection and are often marketed in security and military contexts. It’s no surprise then that they’re often seen in spy movies.

Standard design parabolic mics have limited low-frequency performance because they cannot successfully focus sound with longer wavelengths. For example, the stated frequency range for the Telinga Pro X is 60Hz–20,000kHz. The manufacturers of the Wildtronics Pro Mono, which features integral booster disks, claim a frequency response of 20Hz–20,000kHz. All parabolic mics however are highly sensitive to high-frequency sound. So the question becomes, how important is the capture of low-frequency content for you?



Telinga Parabolic System. Parabolic mics are super directional and highly sensitive to high-frequency sound – but less capable in the low-frequency range.



Wildtronics Pro Mono parabolic mic. Fitted with booster disks to improve low-frequency response.

Contemporary parabolic mics are generally around 30–70cm in diameter and the dish will be made of polycarbonate – vintage models made of aluminium can also occasionally be found on eBay. And there are various enthusiast sites and YouTube clips that offer DIY solutions. These are great fun to try, however dustbin lids and woks will not perform as well as a properly calculated parabola. The larger the dish, the greater the working distance: a 30cm dish will

struggle with sounds more than 18m away, whereas a 60cm dish will function up to roughly 180m. Units designed for sporting applications (such as the Klovar MiK 16 or 26) are solid, rigid structures whereas those designed for birding and general field recording (Wildtronics, Hi-Sound, Telinga) are much lighter with a dish that can be rolled up for easy transport. Most come with high-quality built-in microphones and dedicated mono and stereo units (Hi-Sound) exist. The Klovar Sound Shark is an exception that allows you to use an omni-directional lavalier mic. The Telinga Modular is another exception – allowing you to use your own pencil condenser mics or smaller omni electret capsules for stereo recording. The unit comes with a Rycote Telinga Basket, windjammer and mono/stereo foam cartridges. Whatever you decide, it’s a limited product arena and you’ll be looking at a price range of roughly £300–£2500 depending on your needs and budget.

ENVIRONMENTAL CONSIDERATIONS

Wind Protection

If the field recordist has an arch enemy, it’s the wind. Not the sound of the wind itself; this can be a beautiful, multifaceted thing as it breathes its way across landscapes, through structures and flora. The enemy we are discussing here is the sound created when wind directly hits the capsule of a microphone. This energy, of a much higher magnitude than the subtle sounds we hope to capture, is heard as a rumbling, bumping and buffeting noise. There are very few occasions, even on the calmest days, when a gust of wind won’t come along and mar an otherwise pristine recording. In mild cases it may be possible to filter out the worst effects, but in extreme situations it completely obliterates a recording. Here, there is nothing to be done other than edit around the problem. In some cases, there will only be the briefest of useable passages, even though on the day it seemed relatively calm.



A classic, all-purpose recorder like the Zoom H4n still needs protection to stop the sound of wind hitting the mic.



Windjammers come in all shapes and sizes to suit different microphones and environmental conditions. At a push you can experiment with knitted items like hats, scarves and gloves. Or even a handkerchief!

Many microphones (and hand-held recorders) come with a foam windshield. These are of little use when it comes to wind protection. They can be of use indoors, especially as a form of ‘pop’ shield to manage sibilance when a mic is being used to record voices close up. What is required outdoors is something that will either absorb the energy of the wind before it hits the microphone or something that provides a chamber of still air around the microphone – or both in combination. The basic form of windjammer is made from high-quality artificial fur that absorbs the energy of the wind but has a backing that’s acoustically transparent. These work remarkably well and can deal with average to moderately windy days. They come in all shapes and sizes for different microphones and devices. They’re not expensive, ranging from around £10–£60. Rycote and RØDE are long-established brands offering premium products at reasonable prices. MOVO offers a range of the most cost-effective solutions. For

more extreme conditions, all these brands offer windjammers where the fur is mounted onto a plastic membrane with a hole for the mic so it can be slipped over. This creates a space between the fur cover and the mic itself for additional protection. Here, you should expect to pay between £25–£100.



Rycote double MS Cyclone windshield kit.



Cinela Piano windshield adapted for double MS (PIA-3S).

Finally, for more extreme environments, we have what are called blimp or zeppelin mounts because of their dirigible-like shape. Again, the brands already mentioned can supply the goods in a price bracket between £85–£600. At the upper end of this product arena we also need to introduce Cinela's excellent products, which range in price from around £500–£700. Blimp mounts were originally designed for shotgun mics but there are many adaptations allowing you to use them with a wider range, including stereo mics and micro/small condenser pairs. The microphone is mounted into a shock-absorbing frame that de-couples it from the structure of the blimp to reduce handling noise. This is then inserted into a plastic mesh tube which has a thin layer of foam applied to its surface. In many situations this can be used on its own, but a fur cover offers added protection and there is little acoustic benefit to not using one. Note that both RØDE and Rycote make smaller windshields designed using the same principles as the blimp for smaller mics.

Rycote and Cinela also offer a whole range of custom solutions for specialist mics and multi-mic setups.

Rain

Rain is another factor to be considered when working in the field. For short periods of time in all but torrential conditions I've found basic fur and blimp mounts to offer an adequate degree of protection. The fur cover will absorb the energy of the raindrops falling on it and keep water away from a microphone. There is, however, a point of saturation beyond which the fur becomes less efficient at absorbing energy. And if it gets to the point where the mic capsule itself starts getting wet you will experience noise, crackling, a reduced frequency response, lower output level or a complete loss of functionality. Companies like Rycote and Cinela make dedicated rain covers for their blimps. Rycote has a range of Duck products made from acoustically transparent foam that covers the upper part of the blimp while allowing the fur cover to remain fitted. The company also has a Rainjacket cover for protecting mics between takes. Cinela offers a range of Kelly (*Singin' in the Rain* – get it!) rain covers – again made from foam but designed to replace the fur windjammer cover. These are designed to be used in conjunction with a product called Nikwax, which is used to waterproof the blimp cover itself.



Rycote Duck rainproof cover. While a standard mic blimp will be fine in light rain or drizzle, more thorough protection is required in extreme conditions.



Cinela Kelly rain cover.

Cable connectors also need to be considered if they're being deployed in wet conditions. Most female XLR connectors have a rubber seal ring that makes the joint water resistant but not waterproof. For extreme conditions consider cables featuring Neutrik NC3FXXHD IP67 heavy-duty connectors, which are designed for outdoor use. If you need to join cables in wet conditions, cling film can be used to wrap exposed metal plugs (but not as a long-term solution) or plastic containers can be modified.

If you must record in torrential conditions, try to find shelter such as through an open window or door but beware of the sound of the shelter environment contributing to the recording. For example, rain on a wooden or tin roof creates a very specific keynote sound. Another DIY solution I've seen in the field is to build a shelter for a

blimp under a foam sheet (which will itself absorb rain noise) that's supported by chicken wire, sticks, a spare boom pole or whatever's handy. This works remarkably well.

Recording in the field often calls for improvised solutions, and practitioners often develop their own idiosyncratic solutions. James Bulley, whose approach to multichannel recording we will explore in Chapter 4, has a unique solution for protecting small omnidirectional mics such as the LOM mikroUši in rainforest environments – and this works well in all situations where moisture and the sound of raindrops falling on a mic causes problems. Each mic is protected by a material called rubberised horsehair (often made from coconut fibre these days). It's a model-making product designed for simulating bushes and undergrowth. It's very light, cheap and comes in sheets that can be cut to size. As the name suggests it's quite dense, absorbent and the rough surface absorbs any sound created by drips. It's also very lightweight and can be held in place using wire or crocodile clips. A quick spray with Nikwax makes it completely waterproof. Other weatherproofing solutions he suggests include the use of modified Sistema containers for cable junction boxes, the special plugs and accessories manufactured by Neutrik in its TOP (True Outdoor Protection) range, and Peli Air and Micro cases for storing mics – especially the bright yellow ones as they are easy to keep an eye on.

Temperature Extremes

When working in cold, humid or hot conditions, store mics at room temperature in a container with some small bags of silica gel desiccant (good practice for mic storage generally) and it's advisable to give mics about an hour to adjust/acclimatise to environmental conditions before making important recordings. In extremely cold environments the capsule's diaphragm can lose flexibility and this can lead to noise issues and a change in frequency response. Freezing conditions can also introduce humidity, and therefore further noise issues, into the equation.



Simone Eringfeld on location in Antarctica with electret microphones. Extreme cold can affect the performance of microphones, so monitoring becomes critical.

Another issue to consider in cold climates is effective battery life. All batteries rely upon chemical reactions to function and these slow down as the temperature gets colder. Lithium-ion batteries are less affected than alkaline cells but whatever you are using, endeavour to keep spare batteries in a jacket pocket where body heat can keep them at a good operational temperature until needed. This won't be possible when using battery packs; they need to be fully charged

and kept at room temperature before use and then wrapped in layers of insulation – cloth or even newspaper – before transferring to a kit bag. An insulated picnic cooler box can be useful, as it will help keep them warmer for longer. Maybe toss in a couple of activated hand warmers for good measure.

Microphones are not unduly sensitive to the higher temperatures encountered in natural environments but it is best to avoid leaving them exposed to full sun, especially for long periods of time. Plastics and synthetics do not benefit from prolonged exposure to UV/heat so shade is something to consider even when using a blimp. Many high-temperature environments will also be high humidity, which can be problematic because it can cause changes in the mic capsule's capacitance and hence its response. More expensive field recording mics that use RF technology are much better at handling moisture, humidity and high/low temperature extremes. Rapid heating or cooling of microphones can cause condensation inside the capsule leading to noise issues; popping, fizzing and crackling sounds are telltale signs.

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3
CAPTURING HIDDEN SOUNDS

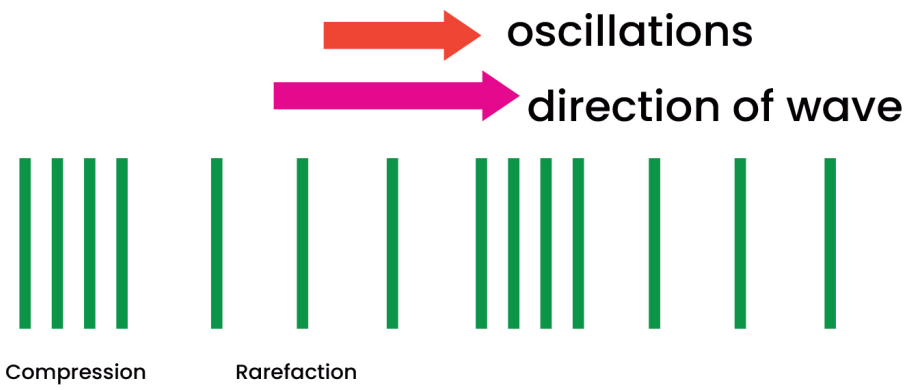
In this chapter we're going to look at a range of very different sounds and environments that all share one thing in common; to capture and explore them, we need specialist microphones or devices. Hence the 'hidden sounds' classification I use. We'll be looking at sound propagating through solids and liquids, sound above or below the range of human hearing and electromagnetic transmissions. For me, this is an area where field recording becomes pure sonic exploration; you simply do not know in advance what you are likely to hear. Sometimes it's familiar sounds, but freshly experienced, as a result of tonal or spatial difference. It can be the sonic signature of fauna that is totally alien to our ears. And it can be iterative electromagnetic cycling, pulsing tones and noises that are more reminiscent of electronic music than natural phenomena. Sound behaves very differently in different mediums so before we move forward with practical considerations, it's worth focusing briefly on sound and transmission itself.



Soma Ether – collecting sound from overhead power lines.

SOME FACTS ABOUT SOUND

Sound is a sequence of pressure waves. It can propagate through gasses, liquids and solids. In gasses and liquids sound is composed of longitudinal (compression/pressure) waves. Sound moves through air compressing and rarefying the gas in the direction of travel (parallel with) of the sound waves as they vibrate back and forth. Only the energy travels. In solids, sound is composed of longitudinal waves and transverse (shear) waves. With transverse waves the particles of the medium vibrate at 90 degrees (perpendicular to) to the direction in which the energy is moving. A ripple on a pond is an easy way to visualise a transverse wave; the water goes up and down while the energy moves left to right, or vice versa. Transverse sound waves in solids usually travel at about half the speed of compression waves and are important in seismology.



The difference between longitudinal and transverse waves.

Waves can be reflected, refracted or attenuated by the medium and all mediums have a number of properties that affect the behaviour of sound propagation. There is a relationship between density and pressure. And this is affected by temperature. These factors determine the speed of sound in the medium. In Chapter 2 we learned that the speed of sound in air is a comparatively leisurely 331m/sec at O°C, rising to 344m/sec at 20°C (air pressure depending). In water, sound waves move at around 1480m/s, which

is approx. 4.3 times faster than in air. Sound transmission through steel happens at around 5130m/s, which is approximately 15 times faster than in air. Another influential factor is the motion of the medium itself, for example the effect of wind, current and so on. This is independent of the motion of sound through the medium: if the medium is moving, sound is further transported.

The next factor to consider is viscosity; this determines the rate at which sound is attenuated – losing signal strength over distance and also frequency content because attenuation is frequency dependent. In air or water, attenuation is negligible. Different materials have different attenuation coefficients, and this is important in materials science. For field recordists, it's useful to know some broad facts, as it will enable us to understand better the tools at our disposal and the outcomes we are hearing.

Electromagnetic waves (including light and radio waves) are purely transverse waves that travel at the speed of light, which is a little less than 300,000km per second in a vacuum. Electromagnetic waves travel very close to the speed of light in Earth's atmosphere.

DIFFERENT TYPE OF MICROPHONES AND OTHER DEVICES

Contact Microphones

These are piezoelectric (*see* Factfile 3.1) transducers used for recording sound waves travelling through objects and structures. They perform best when applied to resonant structures such as metal railings, staircases, boat rigging, wire fences, vehicles and other machines, water tanks and thin sheet materials. Indeed, if you have an acoustic musical instrument with a built-in pickup it will most likely be a piezo transducer. If you do a quick search for musical instrument ‘bug’ pickups you will find lots of low-cost (under £20) contact microphones that can be repurposed for field recording. For a more professional option JRF Microphones offer a standard contact mic for £28 including postage (£39 for shipping outside the UK). These come with a 2m cable as standard and you can request extra for an upcharge. They’re available with either ¼in plugs or 3.5mm mini-jack. The higher-specification C-Series Pro contact mic is also available for £45 (£50 for shipping outside the UK). Again, these come with 2m standard cable.



Recording sound transmitted through metal railings with contact mics. Structures activated by wind, water or human movement can sound amazing.



JRF contact mic. Designed to be placed on surfaces to pick up sound transmission.

For the best frequency response, contact mics need to be used with a buffer preamp such as those made and supplied by Zeppelin Design Labs (US), Aquarian Audio (US) or Stompville (UK). Alternatively, a simple XLR impedance matching adaptor will do the job – JRF can supply these for £25, alternatively, Hosa MIT-129 adaptors can be found for roughly the same price elsewhere online. An alternative is to purchase a contact mic that comes with a preamp, such as the Zeppelin Design Lab's Cortado MkIII ($160) or Michael Krzyzaniak elegant, rechargeable Metal Marshmallow II (£126). There is also the excellent Barcus Berry 4000 Planar Wave System (£440). This is designed for piano and harp, comes with its own preamp and can be repurposed for sound design purposes. However, you don't need to spend this much to explore contact microphones.

You can also make your own contact mics; piezo elements can be purchased from most electrical component suppliers and general online retailers such as Amazon. Multiple disks can be purchased for

under £10 – note that these are often referred to, and sold as, buzzer elements; if you send an electric current to them they will buzz or function as small speakers – this is their main application. For our purposes, disks of 35mm or larger are best and only basic soldering skills, tools and materials are needed to connect them to a cable/plug. For this one quick solution is to repurpose a standard musical instrument cable. Cutting one of these in half will give you cables and plugs to make two contact mics. Once made up, a couple of coats of Plasti Dip will protect them nicely. If you want to get creative and add a little weight, fill a plastic juice or milk carton lid with Femo or DAS modelling clay and press the connected piezo element (you'll need to cut a nick in the side of the lid for the cable) into the wet surface. Once dry, give it the Plasti Dip treatment for a professional finish. These will function both as a contact mike and a DIY hydrophone for underwater recording. Again, for best results you will still need an impedance matching preamp or adaptor.



Contact mics are easy to make: solder the piezo element (right) to an old guitar cable and give it a coat of Plasti Dip.

A contact mic can be used in various ways. One of the most common is to simply put it on a range of surfaces and hold it in place with finger pressure. They can also be attached using small G-

clamps, spring clips, clothes pegs or, at a push, electrical tape. Don’t use Blu Tac or similar as this compromises the mic’s surface-to-surface contact. Not all surfaces will yield interesting results but sooner or later you’ll hit on some magical sounds. The best results come from environments where the material is being activated by external force such as wind, water, mechanical action or the movement of people. Long wire fences can positively sing in the breeze and the type of metal fencing used around building sites can rattle and clang in extraordinary ways. Metal staircases and railings can reveal bell-like tones and airborne sound can get into the fabric of hollow structures so it’s possible to pick up ghostly voices and other sounds as well.

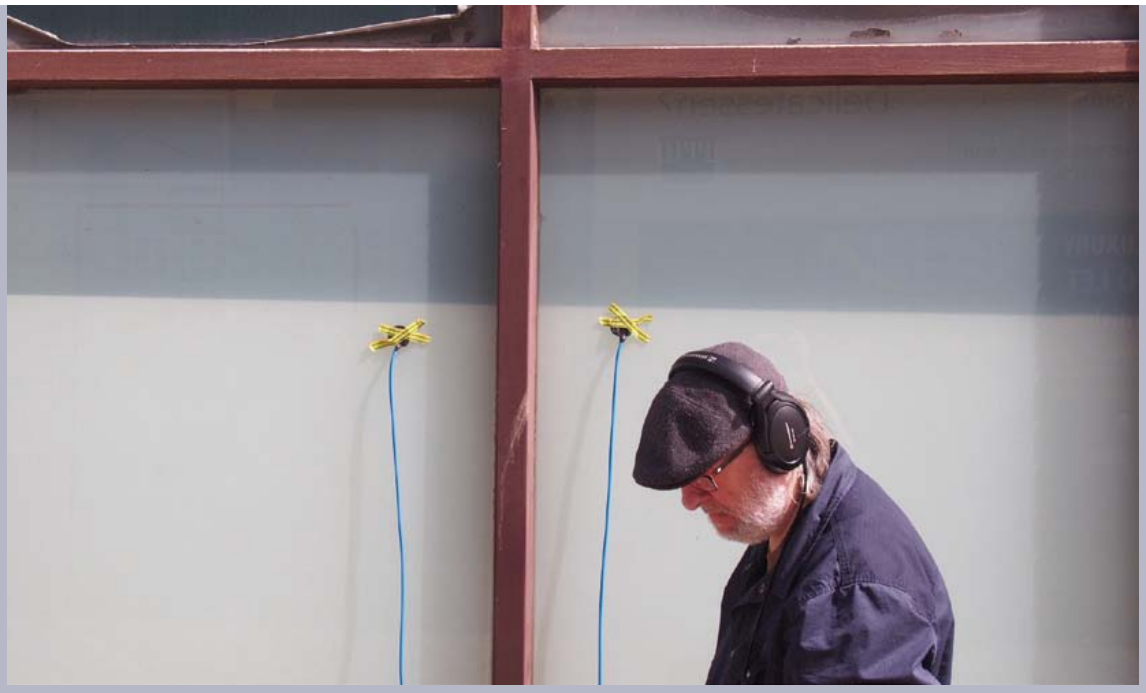
Recording the Sounds of Plants and Trees

Over the years many students have asked me how to record the mysterious inner world of plants and trees. If you have small sub-miniature mics like DPA 4060s these can often be inserted into openings larger than 5mm. The LOM Geofón (*see* later in this chapter) has an attachment that can be pushed into cracks and crevices that allow around 6mm access. However, Jez riley French has come up with a new contact mic adaption: the JRF C-Series+ 'Ecoutic' model (£60). This is designed for listening to plants, soil and other subsurface environments. Each unit is supplied with two detachable probes, and there is a choice of small, medium and long probes (spares are also available for an additional cost). These needle-like probes are easy to insert and will reveal remarkable hidden worlds, capturing the sound of sap moving through structures, insect life, water movement and other hidden sounds.



JRF C-Series+ ‘Ecoutic’ model. Amazing sound worlds can be revealed when the probe is inserted into trees, plants, soil and other structures.

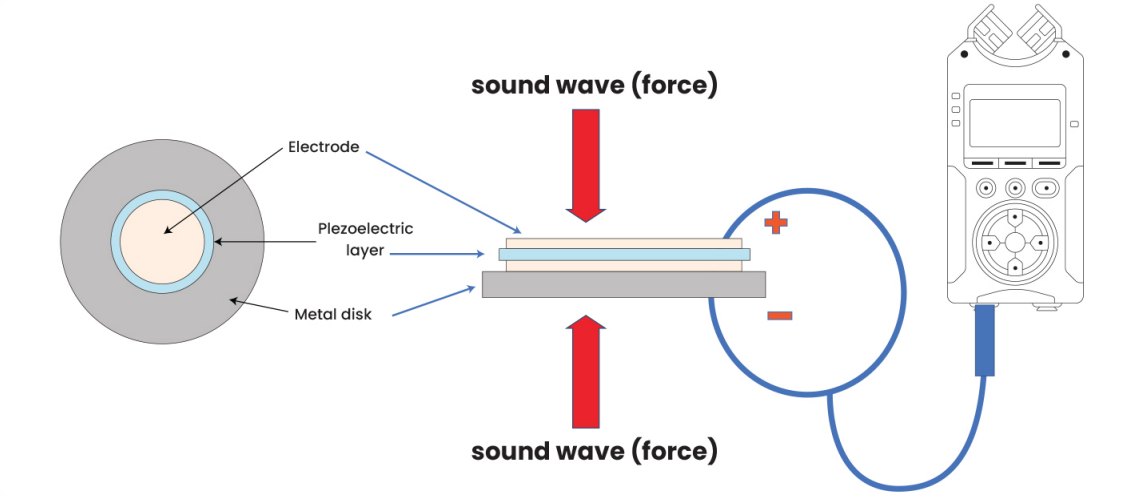
Jez riley French doesn’t refer to himself as a field recordist; he feels the term can now be quite loaded, conjuring up an idea that it’s all about the technology and, too often, set ideas of ‘nature sound’ (Leadley, 2023). He prefers to identify as an artist, musician and composer for whom the recording of locations is only one method of documenting experience, so listening, memory and text all have a role to play. While using a wide range of approaches he is particularly known for his use of contact microphones, which he started using as a teenager when he built a piezo pickup to amplify a zither. Generally unimpressed with the quality and cost of both contact mics and hydrophones he started to improve on his own DIY designs, which led him to start a small-scale business (JRF Microphones) in the early 2000s, based on handmade items. As a creative artist his work is diverse, perhaps because his process is very spontaneous and he’s not motivated to record specific environments. His principal motivation is in the listening. French highlights the importance of detail and scale within recordings as both giving work depth and as the mechanism that extends the range of listening, opening the ears to a wider understanding of the world.



Jez riley French recording with contact mics of his own making. (Photo: Phoebe Riley Law)

FACTFILE 3.1: PIEZOELECTRIC EFFECT

Certain natural materials, such as quartz or lithium sulphate, can directly convert mechanical signals (force, pressure, strain or acceleration) into electrical energy. The effect is more pronounced in a synthetic ferroelectric ceramic such as barium titanate. A contact mic element has a metal support disk and a ceramic element. This converts the energy from sound waves into electrical energy and this can be recorded. Jacques and Pierre Currie first demonstrated the piezoelectric effect in the 1880s. Other uses include record player cartridges, door buzzers, cigarette lighters, ultrasound transducers, television remotes and car sensors.



Piezo element used as contact microphone.

FACTFILE 3.2: IMPEDANCE

Impedance, unlike ordinary DC resistance, is frequency dependent because of the elements in a circuit that may have a reactance – like inductors and capacitors. Reactance is the strict AC equivalent of DC resistance and impedance depends on both DC resistance and the reactance of circuit elements. There are various formulae for calculating impedance for a given configuration of resistors, capacitors and inductors you can find on the web. As impedance is frequency dependent, so its magnitude will change depending on the frequency of the signal.

When it comes to recording devices and microphones, impedance matching isn't always essential as any power losses can be made up by modern preamps. As mentioned above, the impedance depends on both the normal resistance R and reactance of a circuit. If you make R large enough it makes the frequency dependence less of an issue. So, for example, let's consider a preamp with a 10k input impedance, most of which comes from ordinary DC resistance (not reactance) in its design and a 600 Ohm mic connected to it. Here, there is a mismatch of about 10:1 so there will be power losses – but the preamp voltage gain can make up for it. Since most of the 10k impedance comes from ordinary DC resistance, the frequency dependence (which you don't really want) is minimal. However, the piezo elements used in contact mics and some hydrophones have huge impedance compared to a regular mic so they cannot be plugged directly into a preamp without experiencing significant frequency loss; the mismatch results in an unintentional high-pass filter. You won't have a problem with signal level but you'll get a very poor low-frequency response leading to unnatural 'tinny' sounding recordings.

Hydrophones

Sound travels through water at around 1480m/s, which is approximately 4.3 times faster than through air. As a consequence, underwater soundscapes offer rich listening potential; airborne sound is carried into water where it mingles with the sound generated by boats and other craft, buoys and people and with more general underwater sounds associated with the movement of the medium: wave action, sloshing, gurgling and so on. In the oceans, the presence of marine mammals can be detected by the sounds they use to communicate – often over great distances. In rivers and ponds, we hear the sounds of fish and insects. Swimming pools, aquariums and water tanks can offer rich acoustic rewards. Hydrophones can also be used in a more general creative way to record the sounds of objects being dropped into water. The sound of resonant objects such as bells will change in pitch as they are lowered into water and this can be used to create fascinating effects. However, it’s not the case that every hydrophone recording will yield great results; sometimes it can be very quiet down there!



Sound fishing with a pair of JRF hydrophones. You need to get them to hang in free water to get the best results. Hang from a bridge, jetty or use poles as in the picture.



Hydrophone element inside waterproof enclosure.

Many hydrophones rely on some form of piezoelectric technology similar to that used in contact mics. Indeed, a contact mic element

with a waterproof coating will record underwater sound. Purpose-built hydrophones have much better containment and will be more durable. They are also supplied with much longer cables. In terms of best practice, you want to deploy the hydrophone in free water – so not against a riverbank or resting on the river or seabed (unless weighted). In both these cases the movement of the water will cause the cable/mic to move and you'll pick up the sound of them rubbing against surfaces. This is even more impactful than the sound of wind on a conventional mic. Hydrophones work very well when lowered off a pier or bridge. For river and pond recording I have some fibreglass poles (repurposed from a broken garden badminton set!) to get the hydrophones away from the bank. Due to the speed of sound in water, phase issues are less of a problem than with airborne sound – hydrophones can be a couple of metres or more apart in free water without any obvious issue.



Sound fishing in an urban pond. An amazing world awaits: insects, fish and more.

Hydrophones (other than home-made ones) used to be extremely expensive because the only available options were designed for scientific monitoring. These days a JRF hydrophone will cost between £50–£65 depending on whether you choose the standard or higher specification D-Series unit. Both come with 5m cable as standard and more can be requested as a custom order. Monkey Sound offer the H27M hydrophone based on similar design principles for around £70. The company also has an interesting

stereo unit, the H27S for around £90. With all these units you will need impedance matching adaptors or a buffered preamp to get the best, full-frequency results.



Sound fishing – while it's often difficult to get good results by simply casting a hydrophone into a river, the sound of boats, swimmers and marine animals can be recorded if present.



Aquarian Audio A5.



Pictured with 3m cable and XLR output

Aquarian Audio H2d.

Hydrophones are used for many purposes including leak detection and underwater sound measurements in marine, industrial and scientific environments. Hydrophone arrays, where individual units may be thousands of miles apart, can be used to track fish movements. Building long-term data sets of ocean acoustics enables the identification and assessment of the sonic impact of human activity and natural processes on the marine environment. Aquarian Audio is a specialist manufacturer of hydrophones, producing ‘Audio’

and ‘Scientific’ branded products. Though any hydrophone can be used for field recording, the Aquarian Audio brand caters to field recordists, with the goal of making high-quality, rugged recording tools at an affordable price. The H2 has been in production since 2005, with the latest ‘d’ version offering the best sensitivity and lowest self-noise. The latest addition to the Aquarian Audio line is the A5, which is smaller and more robust, and improves upon the bandwidth of the H2, as well as offering internal PIP signal conditioning that drives long cables well and offers stable gain, regardless of current supplied through the mic preamp. The PIP-XLR balanced line driver is added to the A5 for direct connection to mic preamps with phantom power on XLR inputs. All Aquarian’s hydrophones can be ordered with any length of cable.



Dodotronic Hydromic.

One last device to mention is the Dodotronic Hydromic 384K (approximately €2000). This is an audio and ultrasonic hydrophone with its own recording capabilities; it’s designed to be left unattended. The Dodotronic Hydromic 384K USB is an integrated hydrophone and ultrasound mic designed for direct connection to a laptop or smart phone.

Jana Winderen began field recording in 1992 while studying. She started with river sounds and underground spaces, discovering contact mics and hydrophones in the process. She became progressively more interested in sounds that people do not notice or do not perceive and frequency ranges inaudible to the human ear. She has recorded in glacial melt water, under the Greenland ice sheet and captured the sounds of whales, wild cod, toadfish, sea snails, sea urchins, Nordic shrimps and other sea creatures.



Jana Winderen recording location sounds for the installation: *The Art of Listening: Underwater* at Lenfest Center for the Arts, Colombia University, New York. (Photo: Gavin Browning)

Winderen doesn't employ a lot of sound treatments (such as plug-ins) in post-production, preferring creative approaches to recording in the field to get the results she wants. Her work can feature the layering of sound, sometimes including above and below water elements to give a sense of greater perspective. Winderen presents her work in the form of installations, such as *The Art of Listening: Underwater* at Lenfest Center for the Arts, Colombia University, New York; *Listening through the Dead Zones* for IHME, Helsinki and *Rising Tide* at Kunstnernes Hus, in Oslo. She also releases albums and gives illustrated talks.

Her audio-visual work is published by Touch, and her Bandcamp page and YouTube channel provide plenty of listening opportunities. She also has an excellent artist's website.

Coil Microphones and Other Electromagnetic Sensing Devices

The world we live in is bathed in electromagnetic radiation. While there are plenty of fascinating natural sources to explore, contemporary technology is the main source. Computers, monitors, domestic appliances, lights, cabling and sensors fill our homes with it. Underground power cables, traffic lights, cash point machines and car charging points create an alternative map of the urban landscape. Transit systems, industrial and scientific environments are full of bespoke equipment that emanates unique electromagnetic signatures. The air is full of radio waves. None of this can we detect directly because we do not have an appropriate sense organ. However, it’s possible to convert electromagnetic radiation into sound very easily. The radio receiver is one obvious example where the facility is exploited directly for communication and entertainment purposes. There is also a range of low-cost solutions that enable us to explore the electromagnetic realm around us – and the results are often surprising. Expect hums, buzzes, crackles, curious iterative cycles, mixed frequency soundscapes, pulsations, squeaks and squeals. People often liken the sounds to insect recordings or electronic music and much of it, by association, has a dystopian feel or gestures at science fiction soundtracks.



Exploring the electromagnetic sound of a parking meter using LOM Elektrouši stereo coil mics. Stereo recording is particularly rewarding because sounds are very localised – so you can ‘play’ a cash point machine for example.



A basic telephone bug pickup is all you need to collect surprising electromagnetic sounds generated by every day devices.

One of the most immediate and cost-effective solutions for listening to electromagnetic radiation is to repurpose a bit of readily available old technology; the analogue telephone ‘bug’ pickup. These usually have a suction cup attached and were designed to be stuck to the back of an analogue handset to record conversations. They are readily available online and cost between £1.50 and £10 – and the cheapest work, for our purposes, as well as the more expensive.

They work via magnetic induction in a small, closed coil of wire. Faraday's Law of Induction states that a changing magnetic field causes an electric field to be set up in a nearby wire, causing a current to flow if the wire is part of a closed circuit (a loop). Any coil of wire will do: solder a cable to the terminals of an old speaker coil and you have an effective coil mic – this will work as well as a telephone bug pickup – but not necessarily better and it will definitely be much heavier! All you have to do is plug the bug pickup into your recorder and monitor with headphones while you explore and record. Then it's a case of passing the bug over different surfaces, electronic and electrical devices. Sounds are usually quite localised so a few centimetres can make a lot of difference. For extraordinary stereo effects, use two bugs at the same time.

There's a range of purpose-built electromagnetic listening/recording devices that offer a wider frequency response or higher gain output than conventional bug pickups. LOM make several. The Elektroucho Pro (€33) is a mono, passive sensor that's significantly more sensitive than a telephone pickup coil. The Elektrouši (€30) is a useful stereo pair. The company also makes the Priezor (€110), a very sensitive passive magnetic antenna for electromagnetic listening that's capable of capturing faint atmospheric events in the VLF 'natural radio' bands.



LOM Priezor – another device designed for capturing airborne electromagnetic signals.

LOM's Elektrosluch 3+ (€100) is a powered device for EMF capture. The signal-to-noise ratio is very good and there is a gain boost control for quiet signals. It's the most sensitive device LOM makes and has an extended range in terms of both high- and low-frequency response. This will allow you to capture even more delicate electromagnetic sound in greater detail and definition. LOM also make the Elektrosluch Mini City (€50), a DIY self-assembly kit if you're a bit handy with a soldering iron.

Soma Laboratory's Ether (£120) is another device for electromagnetic listening and recording. The company refers to it as a kind of anti-radio. Rather than contain a tuning circuit to focus in on a particular station, it samples the entire spectrum. It has a magnetic antenna, like an old-fashioned radio, and antennae printed on the PCB. There are two external antenna pins and touching them to any conductive surface (including yourself) can supercharge the sensitivity of the Ether. While the Ether will detect the close-up magnetic fields like a bug-style pickup, a bug is preferable for 'cleaner', more focused electromagnetic recordings. Where the Ether excels is in monitoring general environments, especially away from the built environment where bugs can pick up very little. Keep all phones and recorders well away unless you want to pick up their emanations!



LOM Elektroucho Pro. A sensitive, passive device for collecting faint electromagnetic sound.



LOM Elektrosluch. Highly sensitive device, designed for gathering very quiet electromagnetic signals.



The urban environment is a great source of electromagnetic sound. Cash machines, window displays, charging points and cables running under the street generate an alternative sonic map of the city.



Alan Cook recording with the Soma Ether. Note wrist-mounted Zoom F3: a very practical solution.

For those interested in the most esoteric sounds, it’s time to turn our ears to EMF sources in the natural world. Sferics, the sound of lightning, is created when it travels through the Earth’s magnetic field generating electromagnetic energy. This can be picked up in the Very Low Frequency (VLF) radio band. The frequency range is between 3Hz–3000Hz and can be recorded as sound using a VLF receiver and antenna. A fun alternative is to build a DIY lightning

sound collector. Alvin Lucier and Pauline Oliveros experimented with recording sferics in the 1960s and Lucier released a record (called *Sferics*) in 1981. These crackling sounds are referred to as ‘tweeks’ or ‘bonks’ by scientists. Longer musical sounds with a descending character are called ‘whistlers’. These are caused when sferics travel long distances along the Earth’s magnetic flux lines. Whistlers can last for several seconds and the apparent pitch change is the result of frequency dispersion due to the distance travelled. Huge distances are required to generate large dispersions.

There is a further, rare type of sound referred to as ‘dawn chorus’ as a result of its similarity to a flock of birds. To recreate Lucier’s style of antenna you’ll need a couple of pieces of timber, as long as practical, and plenty of copper wire. The wood is made into a cross and the wire is wound around the edges of the ‘X’ to create the large loop antenna. This simply needs to be connected to the input of a sound recording device. For best results, station your antenna on a hillside well away from power lines when the right climatic conditions are imminent. For more details on antenna construction, see *Listening to Radio Energy, Light and Sound* (1978) by Calvin Graff.

Sferics fall into a category of sound sometimes referred to as natural radio and realistically, you’ll need more advanced VLF listening equipment to explore this aspect of the magnetosphere fully. However, with a suitable receiver you may be able to capture the sonic traces of disturbances originating from the solar winds and geomagnetic storms. These are most likely to be heard in the hours around sunrise. Meteor showers scatter dust and debris that ignite on contact with our atmosphere and this leaves a trail of ionisation in the upper atmosphere that can scatter and reflect radio waves. During a meteor shower your receiver may be able to pick up brief echoes of distant radio signals that are normally out of range.

Christina Kubisch is a sound artist, performance artist and composer. Much of her work is a synthesis between audio and visual outputs. She began field recording in the early 1970s, graduating from a cassette recorder to a Uher and used this material in compositions. During the late 1970s she studied electronics and through this became aware of electromagnetic induction. Her first installation featured the sound of waves and breath, and participants could use small hand-held induction amplifiers to mix the sound sources in the air which were transmitted through wires in the space. For installations she then designed special headphones that could convert the ambient electromagnetic radiation into sound. These also worked effectively with the sound given off by technology in public spaces such as cash point machines, power cables and lights and they could be adapted for recording. In 2003 she began a project entitled *Electrical Walks* as both a private and group exploration of this urban electromagnetic soundscape. As a composer she works extensively with the electromagnetic sound material she collects and the more casual recordings she makes while travelling around the world. Her method of recording is quite unusual because she likes to move microphones and electromagnetic detectors about so as to compose with sound as it is recorded rather than treat sounds in post-production, so body movement plays an important part in the ‘mixing’ of sounds.



Christina Kubisch electromagnetically recording in Prague at the underground power station (2020).

Ultrasound Detectors

The range of human hearing is generally considered to be 20Hz–20kHz. However, there is plenty of sound outside this range, with content above 20kHz being referred to as ultrasound. For example, bats use it for echolocation to determine the size and shape of objects in the environment and for hunting flying insects. Rodents use ultrasound for communication. When a dog gets excited immediately before the return of its owner it's not some sixth sense, it's because they have picked up their 'ultrasound signature'; as we move, the money in our pockets, belt buckles, keys and shoes all give off ultrasound. Anything with moving parts – bicycles, cars, boats, trains – gives off ultrasound, as does some contemporary electrical equipment. LED lighting is a particularly rich source, caused by components used in driver and dimmer circuits. Many technologies use ultrasound for detection of objects, measuring distances, cleaning, medical imaging, therapy and motion detection. There is little public awareness of ultrasound noise pollution simply because humans cannot hear it. However, a growing body of evidence suggests that high-level, high-frequency ultrasound can cause fatigue and nausea. In addition, intense long-term exposure to ultrasound may potentially affect cellular development.



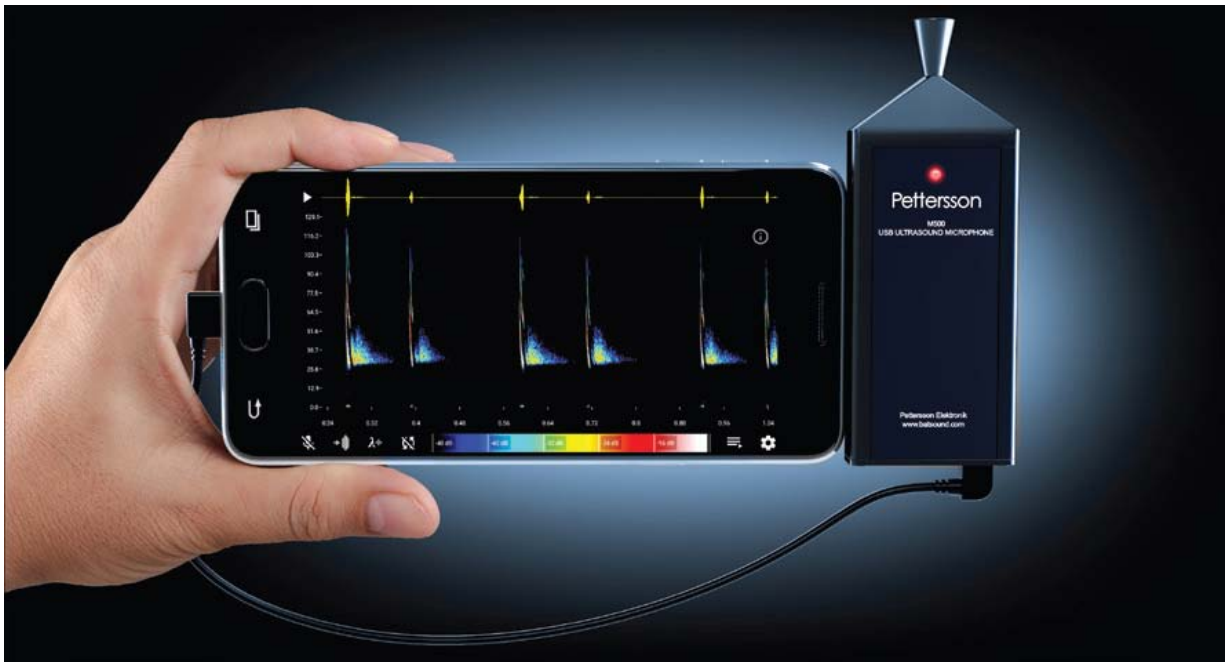
A bat detector will also pick up ultrasound generated by components used in fluorescent and LED lighting systems.



Dodotronic Ultramic UM384BLE.



Dodotronic Ultramic UM384KEVO.



Pettersson M500-384 USB Ultrasound Microphone.

Just because we cannot hear ultrasound doesn’t mean that a microphone cannot record it. Microphones using electret capsules and MEMS (micro electro mechanical systems) technology are particularly good. A lot of general marketing material will tell you that a microphone’s frequency response range is 20Hz–20kHz simply because this is the assurance most people need. It doesn’t necessarily mean they aren’t capable of capturing higher-frequency sound. For example, the EM172, 258 and 272 electret capsules used by LOM and FEL MicBooster have a usable response into the 40kHz range and LOM also has a specific ultrasound mic, the Clippy Ultra XLR, for around £70. The Knowles SPU1410 (MEMS) capsule goes much higher, reaching to approximately 100kHz, albeit with much-reduced sensitivity in the upper reaches. Dodotronic make a range of USB ultrasound mics such as the Ultramic UM192K/200K/250K and 384K EVO (€200–€250) which connects directly to a Mac, PC, iOS or Android device. There’s also the 384K BLE (€350) which features a built-in micro-SD recorder. The Pettersson M500-384 USB Ultrasound Microphone (£440) is another excellent example. It’s a USB device enabling recording to smart

phone or laptop and can be used with a wide range of operating systems.

There's also another, very immediate, way to engage with ultrasound in the field: a bat detector. Although packaged and marketed for the identification of different species of bat, these hand-held devices can be used to listen to, and record, all forms of ultrasound in real time – or almost real time. Lower-cost units all have some form of scanning control that lets you 'tune-in' on different ultrasound frequencies. This does mean they are only sensitive to a limited range of frequencies at any one time. The frequency of the incoming ultrasound signal is combined with a second signal of a similar wavelength supplied by an internal oscillator. This creates a beat frequency that is called a heterodyne. There are actually two frequencies generated (sum and difference) and the one within the range of human hearing is fed to the headphone/recording output.

There is a different type of detector that works by a process called frequency division. Here, the entire ultrasound range is sampled continuously and transformed into the range of human hearing in real time. This is achieved by counting the number of cycles of the ultrasonic signal and generating an audible output signal. Frequency Division detectors are only capable of tracking one frequency at a time, so the output can't be used for harmonic analysis. The final (and most expensive) type uses time expansion detection – again, sampling the entire ultrasound range all the time. These units record a signal in digital memory and then play it back at a slower speed. The delay between recording and playback is minimised to around 100msec, hence playback is an 'almost' real time experience. This creates by far the best audio output and recordings produce highest-quality sonograms. Most high-end units offer multiple detection methods so attaching a recorder to the stereo output enables you to record the heterodyne sound on one channel, and the time-expanded sound on the other.



Listening for bats in the twilight. A stereo bat detector is best because you can record and follow their motion as they fly around.

The NHBS (Natural History Book Service) online equipment shop is a great place to browse a range of bat detectors (and other cool stuff!) and has far more information than the average online retailer. Unless you're interested in the serious study of bats and conducting long-term research, most of the more expensive options with recording functions, built-in GPS or designed for long-term installation will be unnecessary. So, a budget between £20 and £300

will provide what’s required. At the lower end of the price range there are a number of DIY kits for building basic units. While the Haynes HBD2766 kits only require push connectors and assembly, the Wadda WSAK8118 requires more advanced soldering skills. This however is a stereo detector – more about this in a moment. A basic bat detector such as the Magenta Bat 4 (£75) has a rotary dial for frequency adjustment. For greater accuracy, the Magenta Bat 5 (£105) has a digital frequency readout.



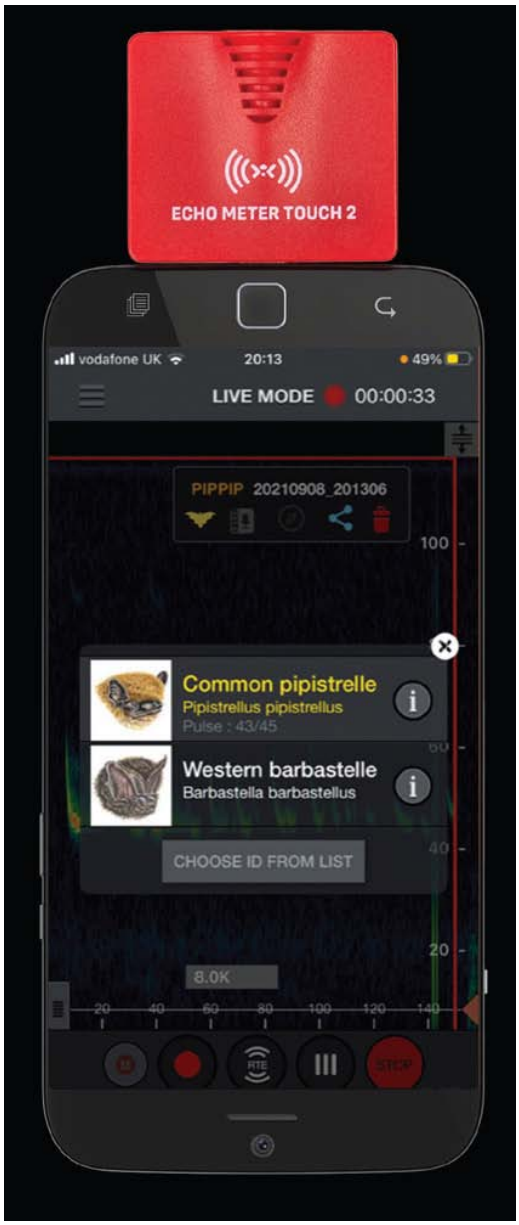
Magenta Bat 4. A good basic bat detector.



Magenta Bat 5. A digital frequency meter lets you more precisely identify the frequency of the bat sound – and this enables you to identify the species.



Pettersson D-240 Bat Detector.



Echo Meter Touch 2.

There are various units, such as the Pettersson D-240 Bat Detector (£300) offering enhanced features such as lower self-noise output or the Elekon Batscanner Bat Detector (£250) with manual/auto frequency detection. A stereo version is available for around £300. I find stereo ultrasound detectors far more useful than mono ones because they can provide movement information for the sound source. There's also Wildlife Acoustics' Echo Meter Touch 2 (£200) solution for Android phones and tablets. It's a plug-in (USB-c)

ultrasound mic that works with a free app. You can listen to calls, view sonograms and record audio. If your device has GPS functionality you can tag calls and display them on Google Maps.

In use, these scanners are simple to operate and all can ‘sniff’ out ultrasound from a wide range of sources, not just bats. I find manual frequency scanning to be more rewarding than automatic for general investigations as it allows you to focus in on specifics that aren’t necessarily the dominant source. Many detectors have a small onboard speaker as well as a headphone port for group listening – for recording, simply connect the headphone port to a recorder’s input using an auxiliary cable and monitor through the recorder.

Recording Infrasound

Now let’s consider sound below 20hz; there’s plenty down there. While our ears may not hear it, other parts of our bodies can sense infrasound vibrations. Just like ultrasound, long-term exposure can adversely affect us, potentially causing nausea, fatigue and sleep disturbances. Infrasound has also been implicated as a possible cause of the ‘spooky’ sensations associated with ‘haunted’ houses. Earthquakes, avalanches, volcanoes, waterfalls and lightning all generate infrasound. Animals such as whales, elephants and giraffes use it for communication over long distances: infrasound can traverse great distances and pass though solid structures (including the planet) without significant dissipation. Human-made sources of infrasound include sonic booms and explosions, heavy machinery, slow rotation motors and wind turbines.



LOM Geofón in use on beach. Sound is transmitted through any material or structure that animals or people move about on.

Microphones can capture sound below the 20hz threshold of human hearing but only with significant modification. Electret condensers can be used, but the capsule itself has to be modified by removing the plastic cover and directly attaching the microphone head to a vibration surface. Thankfully electrets are relatively low cost. In experimental situations the lower limit of the frequency response can be as small as 0.5Hz. If anyone is interested in pursuing this line of enquiry there is an interesting paper, *Response Identification in the Extremely Low Frequency Region of an Electret Condenser Microphone* by Yih-Nen Jeng, Tzung-Ming Yang and Shang-Yin Lee, which is available via MDPI or ResearchGate.



LOM Geofón with spike attachment being used to record internal sound of a tree. At the right time of year you can hear fluid movement, insect sounds and creaking sounds.

Another way to record low-frequency sound is to repurpose a low impedance analogue seismic sensor or device designed for detecting vibration in large machines or bridges. Basic units cost between £20 and £70, with much more expensive, bespoke solutions also being available. The general term for these devices is accelerometer; units that convert movement or vibration into voltage. They're designed to provide signals that can be displayed as visual information and are not ideal for generating an audio output but are capable of providing a voltage feed to the input of a sound recorder. The most convenient solution is LOM's Geofón, a seismic sensor adapted for audio recording with an XLR connector and built into a neat enclosure featuring interchangeable attachment methods: a magnet for metal structures/objects, a suction cup for attachment to glass and a ground spike which can also be used to penetrate trees or other structures. The frequency response is given as 10Hz–1kHz

+, so there is some low-frequency extension but a poor high-frequency response. The Geofón is a surprisingly flexible tool with a number of alternative applications. I've seen them deployed in sound sculptures where they can be an alternative to contact mics and even for experimental drum mic'ing.

More generally, detecting the presence of infrasound and identifying frequency is less of an issue. Phones contain accelerometers and there are apps designed to detect (but not play), display and record infrasound information. A range of apps are available for both Android and Apple devices, such as Sergio Gudkov's Infrasound Detector (Android) and the RedVox Infrasound Recorder (iPhone) for example.

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4
MULTICHANNEL SOUND RECORDING

As a concept, multichannel sound reproduction has been around since the 1940s. Quadraphonic hi-fi systems for the home were an expensive novelty in the 1970s and it wasn't until the 1980s that the general public really started to become aware of the potential of multi-speaker sound through cinema experience. While the growth in home cinema in the 1990s expanded awareness for some, it was not until streaming services began offering Dolby Atmos mixes for music and films in the early 2020s that it became widespread – principally because of the availability of low-cost smart speakers. For the field recordist who's interested in exploring surround sound, now's a great time; the cost of recording equipment has fallen dramatically and contemporary laptops and software can handle multichannel postproduction with ease.



David Monacchi using three simultaneous recording systems (quasi-binaural stereo, first-order ambisonics, high-order ambisonics) in the Amazon, Ecuador, during the Yasunì 2016 expedition. (Photo: Alex D’Emilia)



Barry Truax in the FSU multichannel studio in 1996. (Used by permission of the World Soundscape Project, Simon Fraser University)

How many speakers is a practical minimum for surround reproduction? One model of particular relevance to field recordists is the 8-channel format pioneered at Simon Fraser University (SFU) in British Columbia, Canada in the 1990s. Composers associated with the World Soundscape Project, such as Barry Truax and Hildegard Westerkamp, were interested in exploring the benefits of surround presentation for works based on environmental sound and the 8-speaker multichannel offered many advantages. Arranged in a circle around the composer in the studio or audience at a venue (with speakers/channels numbered clockwise starting front left), there was no obvious centre position (as with 5.1 or 7.1 diffusion), which more accurately reflected the natural aesthetic of environmental soundscapes. Also 8-channel audio was effectively ‘more surround’ than the 4-speaker quad the SFU team started out with – which could be used creatively but was not immersive in the same way.

The SFU 8-channel format established a digital audio workstation (DAW)-based approach and as costs of computer and audio hardware have come down, the format has become one that’s within reach of independent artists and enthusiasts. In addition, 8-channel capability gives you latitude to explore quad, 5 and 7-speaker soundscape composition.

Surround Recording on a Budget: Zoom H2n

For those with only a small budget there are still ways to explore surround recording. Once again, Zoom comes to the rescue with its budget H2n Handy Recorder. This is still a current product in the Zoom range (mid 2025) but will at some point be discontinued as a new 32-bit H2essential has already been released. Note however that this is not a straightforward replacement and no longer offers an onboard 4-channel surround recording mode. However, the Front and Rear recording mode can be used to generate two stereo files that can be correctly angled for spatial audio using stereo plug-ins in a DAW. Looking at the standard H2n, the device format has been around for a while and offers an extraordinary feature set in a very compact, low-cost package: 5 built-in microphones and 4 different recording modes – X/Y, mid/side, 2-channel surround and 4-channel surround.



Zoom H2n. An entry-level device for people wanting to explore multichannel sound recording.

It can be used as a multi-purpose USB microphone and an audio interface. With the 4-channel mode, you can record the signal from the X/Y mics onto one stereo track and the signal from the MS mics onto another. The resulting pairs of tracks can then be routed to separate speakers for an immersive listening experience. Firmware v 2.0 and above gives access to Spatial Audio mode, creating a single 4-channel, multitrack WAV file that includes omni, left/right, and

forward/backward tracks. While this is essentially an Ambisonic file type (more of this later in this chapter) a design limitation of the H2n means it can't record vertical information, hence one of the 4 channels in the multitrack WAV will be empty. The H2n's Spatial Audio files can be combined with 360-degree VR videos recorded using a compatible camera rig. There's an alternative Zoom recorder capable of recording height information: the H3-VR, which we will look at in the next section of this chapter, as it's an Ambisonic recording device.

Microphones

For obvious reasons, recording surround sound in the field with multiple microphones is far more involved and less practical than simply grabbing a shotgun or a pair of omnidirectional mics. There are many surround mic'ing studio setups that can be adapted for outdoor deployment and there are a number of multi-capsule mics designed specifically for field recording. I'm going to focus on these all-in-one solutions. If you're interested in professional studio multi-mic options let me direct you towards the excellent resources on the Los Senderos Studio website: (https://lossenderosstudio.com/article.php?subject=17). However, this doesn't mean that we can't experiment creatively in the field with multiple microphones, especially if we're developing sound for an installation rather than a specific reproduction format. Exactly how many mics and input channels will depend on either how many you have available or the playback format you are working towards. Here I'm talking specifically about channel-based reproduction rather than processed spatial formats. Some basic rules still apply.



5.1 recording using DPA mics and Decca tree mount.

If you’re recording diffuse environmental sound without a clear central focus, multi-capsule Ambisonic mics are very useful and we’ll be focusing specifically on these later in this chapter. Where there is a central focus microphones need to be arranged around the source at the same distance, and ideally the distance between each microphone should be the same so as to take equal advantage of the expanding spherical wave front. However, environments rarely allow for such precise placement so there is usually some level of compromise. Cardioid pattern mics are suitable; these are directional by nature so channel separation is more precise in both the near and diffuse field. However, depending on the microphone, off-axis sound may be tonally coloured. And while sound from the sides and the rear of the microphone is attenuated it will still, to some extent, be present in the recording and may impact authenticity or muddy the end result. Hence mic choice needs to be well researched for your purpose. With omnidirectional mics, off-axis sound is generally more

even and less coloured. While there may be a degree of channel ‘leakage’ this is not such a bad thing, so long as it sounds good. Omnis offer good channel separation in the near field but less precision in the diffuse field, so depending on scale and focus you may want to experiment with different options.

5.1 Recording

Given that 5.1 surround is an established standard, it’s no surprise that there are some all-in-one recording solutions on the market. The DPA 5100 Mobile Surround Microphone (£2800) contains 5 microphones with a break-out cable to 6 XLR connectors (the sixth channel is the LFE output which is derived from a sum of the left and right microphones). Holophone (produced by Rising Sun Productions) also has a number of 5.1 mics; the H3-D (£1600), the H4 Super Mini (£2000) and the H2-PRO 5.1 (£3800) and it also has the H2 pro (£4700), a dedicated 7.1 microphone. As self-contained mics requiring minimal setup, they are ideal for use in the field.



DPA 5100 Mobile Surround Microphone. A very practical mic for 5.1 recording in the field.



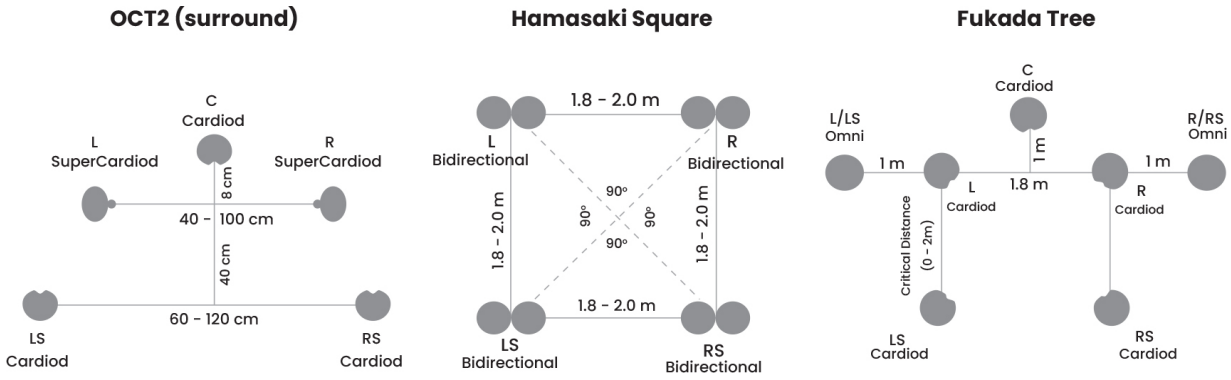
Holophone H3-D.



Holophone H4 Super Mini.



DPA 5006A mic kit. A matched set of high- quality mics for location 5.1 recording.



Three different surround mic’ing techniques.



DPA S5 Decca Tree mount. Designed for location 5.1 recording. Mic position is critical and this takes the struggle out of a set up.

If you want to explore surround recording with multiple microphones, DPA has 3 different surround mic kits (5006-11A, 5006A, 5015A) featuring different mics or combinations of mics in the price bracket £7000–£10,000. There is also a lightweight DPA S5 Decca Tree mount (£2300), which standardises mic placement. Correct placement is critical, and it can be adjusted to create a good range of different surround array standards such as OCT2, Hamasaki Square, Fukada Tree and the DPA speciality WCSA (Wide Cardioid Surround Array). This, or a custom build mount, is realistically the only practical solution for translating formalised surround multi-mic'ing techniques into the field.

Multichannel Sound Recording Case Study #1: Sanctuary of the Unseen Forest

A recent collaboration between James Bulley, Andres Roberts and Marshmallow Laser Feast (an artists' collective that creates large-scale audio-visual environments for galleries and museums) represents a good case study in bespoke multi-microphone technique for multichannel field recording. Broadly speaking, *Sanctuary of the Unseen Forest* looks at human perception within the natural world. Visually it combines LIDAR scanning with a photogrammetry process to create a 3D rendering of the ecosystem around a Capinuri (*Maquira coriacea*) tree in the Columbian Amazon. Bulley made a series of spatial recordings over a number of 24-hour periods; 16 LOM mikroUši microphones were arranged (using Joby GorillaPods) around the tree in two circles of 8 mics (roughly 20m in diameter) at different heights. As he points out, creating perfect circles of microphones in the field is rarely possible; natural landscapes nearly always present some form of obstruction.



James Bulley recording for *Sanctuary of the Unseen Forest*. (Photo: Marshmallow Laser Feast)

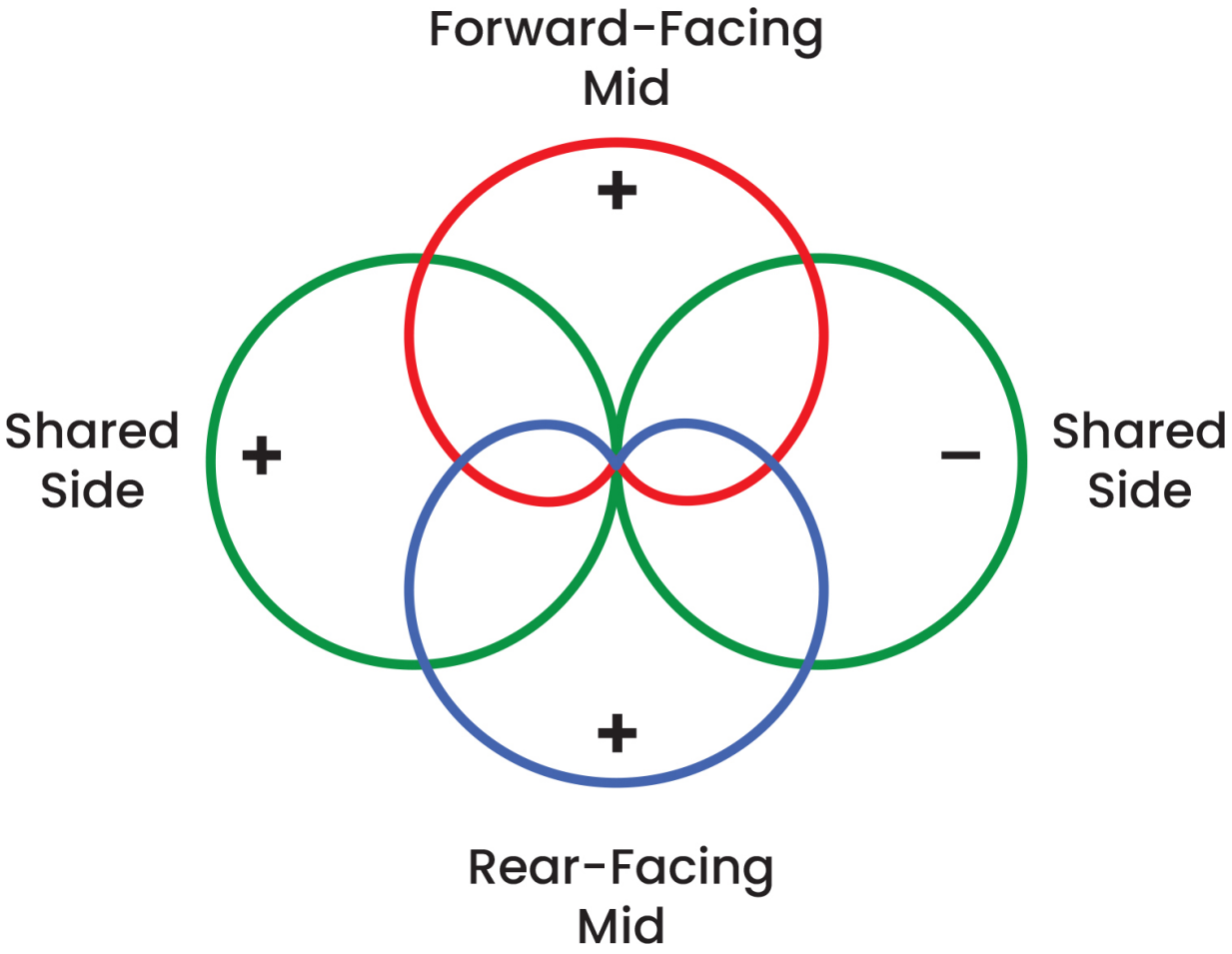
The important thing is to know the distances between the microphones and where they are located. He uses a Bosch laser measure or, if the laser dot is too difficult to see, orange string, which is marked out in metres. This information is combined with pencil drawings, photographs and film documentation of the arrays. Later, in the studio he will work from the principle that each mic position is a speaker position and create a ‘landscape’ that will have the same physical dimensions in the abstract world as the starting point of the mix.

At the same time as setting up the mikroUši array, Bulley installed a Sennheiser Ambeo mic as close to the centre of the array as possible. Having done plenty of A/B testing he generally finds an array of microphones better suits his purpose; he's mostly in the situation of creating big spatial installations in relatively large spaces, so you have an expanded area that reveals a lot of the detail of the forest soundscape. However, the Ambeo provides a useful alternative or supplement. With a single multi-capsule microphone you have a recording from a fixed central perspective.

Once fully installed using lightweight XLR cables with a recorder (time-code synced Sound Devices MixPre-10s with custom sleds for multiple camera batteries) in a waterproof enclose, this 'drop rig' was left unattended so that the environment could settle without human presence. However, this wasn't the end of the process; once he had left the drop rig site, he then set off with his 'roving kit' (Sennheiser MKH40 Shotgun, Aquarian H2 and H4a hydrophones, LOM Geofons and a spaced pair of DPA 4060s on a custom bag rig) to gather more material. So, for example, he was able to record monkeys at dawn, hydrophone recordings of a river with photosynthesising plants and the sound of a rainstorm from under the river – more experimental-focused stuff – while the big core spatial recording was taking place simultaneously elsewhere.

Double M/S

Double M/S also provides a practical solution for 5.1 surround recording in the field because only 3 mics are required, and these can be housed in a conventional blimp windshield. In addition to a front-facing cardioid or supercardioid (mid) microphone and a figure-8 (side) microphone, a rear-facing directional microphone is incorporated into the setup. The front-and rear-facing microphones share the signal of the figure-8 microphone, forming two complete, back-to-back M/S systems. The forward-facing part of the array provides the 3 front channels, while the other provides the 2 surround channels. Schoeps offers two different complete solutions: the Double MS Windshield Set with cardioids (£5040) featuring 2 × CCM 4 L microphones and a CCM 8 L compact microphone (Fig-8), or the CMIT Double M/S Set (£5200) featuring a CMIT 5 U shotgun microphone, 1 × CCM 8Lg figure-8 compact microphone and 1 × CCM 4Lg cardioid compact microphone. Both kits feature extension cables, windshield with fur cover, pistol grip with the necessary mounts and a breakout cable. For flexible processing in post-production there is a free Schoeps Double MS plugin. For people with the necessary mics already, Rycote make a double M/S Cyclone windshield kit (£830).



Polar patterns and mic arrangement for Double MS set up.



Schoeps capsules mounted for double M/S inside Rycote Cyclone windshield. A practical and portable set up for location 5.1 recording.

Field Recorders

As well as microphones you will need a multichannel recording device with at least 4 mic inputs. There are many options at different price points: the Zoom H6essential, H8, F8n Pro, Tascam DR70d or the Sound Devices Mix Pre 6 or 10 for example. Again, it's best practice to link the input channels together (most multitrack recorders have this feature) so a single control will adjust all the levels at the same time. An alternative to a dedicated field recorder is to use a laptop with a USB audio interface. You may already have something suitable if you have an audio workstation setup. Note however that any laptop/interface combination will be less portable than a multitrack recorder and you'll need an interface capable of running on USB bus power (unless you have mains access on location) such as the Tascam US-4x4HR. And the demands of running 4 mic inputs with phantom power will put a significant load on a laptop battery – so you'll need to augment this with something

like one of the Explorer models from the Jackery Portable Power Station Range if you intend to do any serious amount of recording.



Zoom H8 – 6 mic input channels or up to 10 with the optional EXH-8 module featured here. A great multi channel recorder for field work.



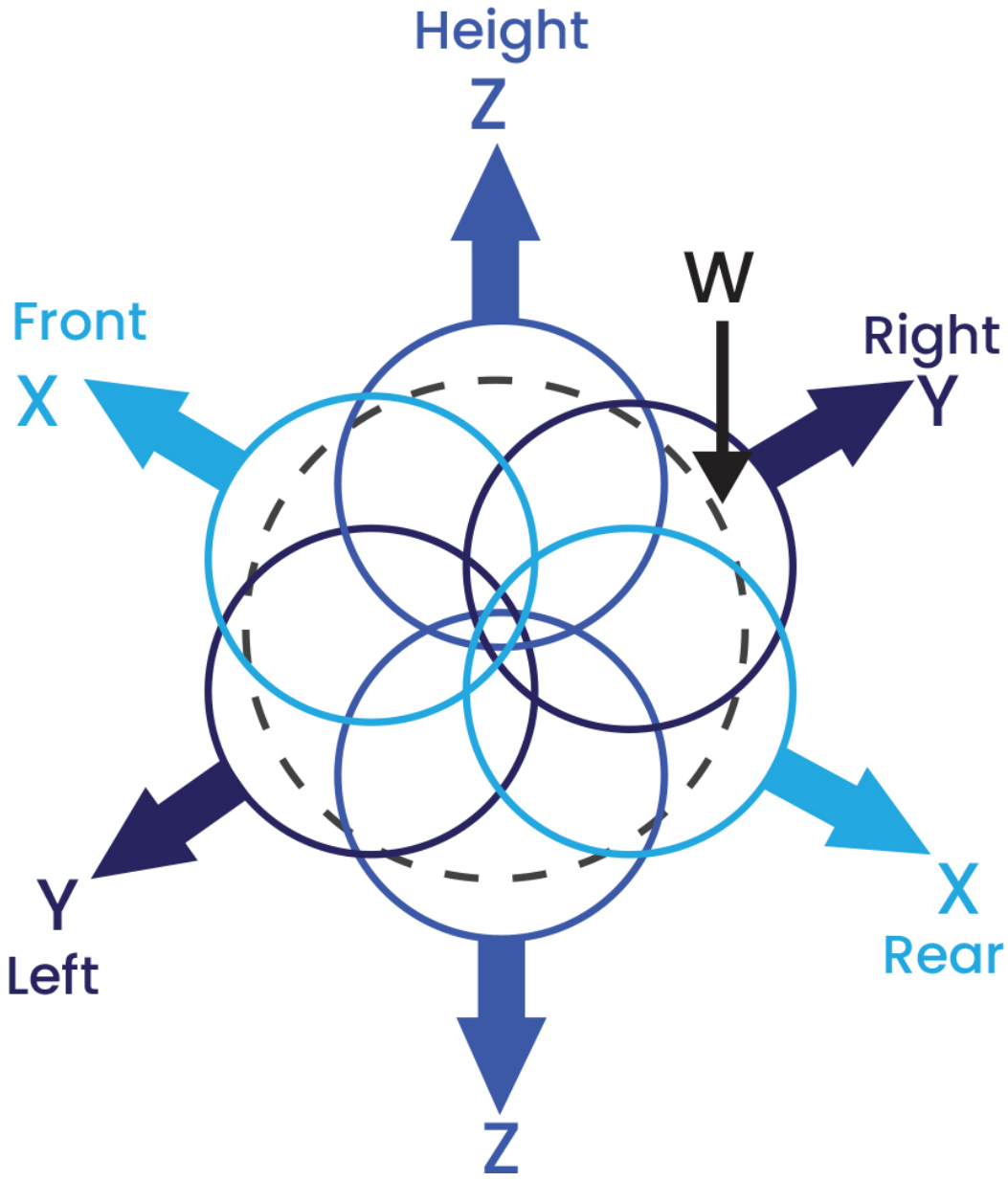
Sound Devices MixPre-10: 8 mic inputs on XLR. Superb sound quality and robust design.

AMBISONICS

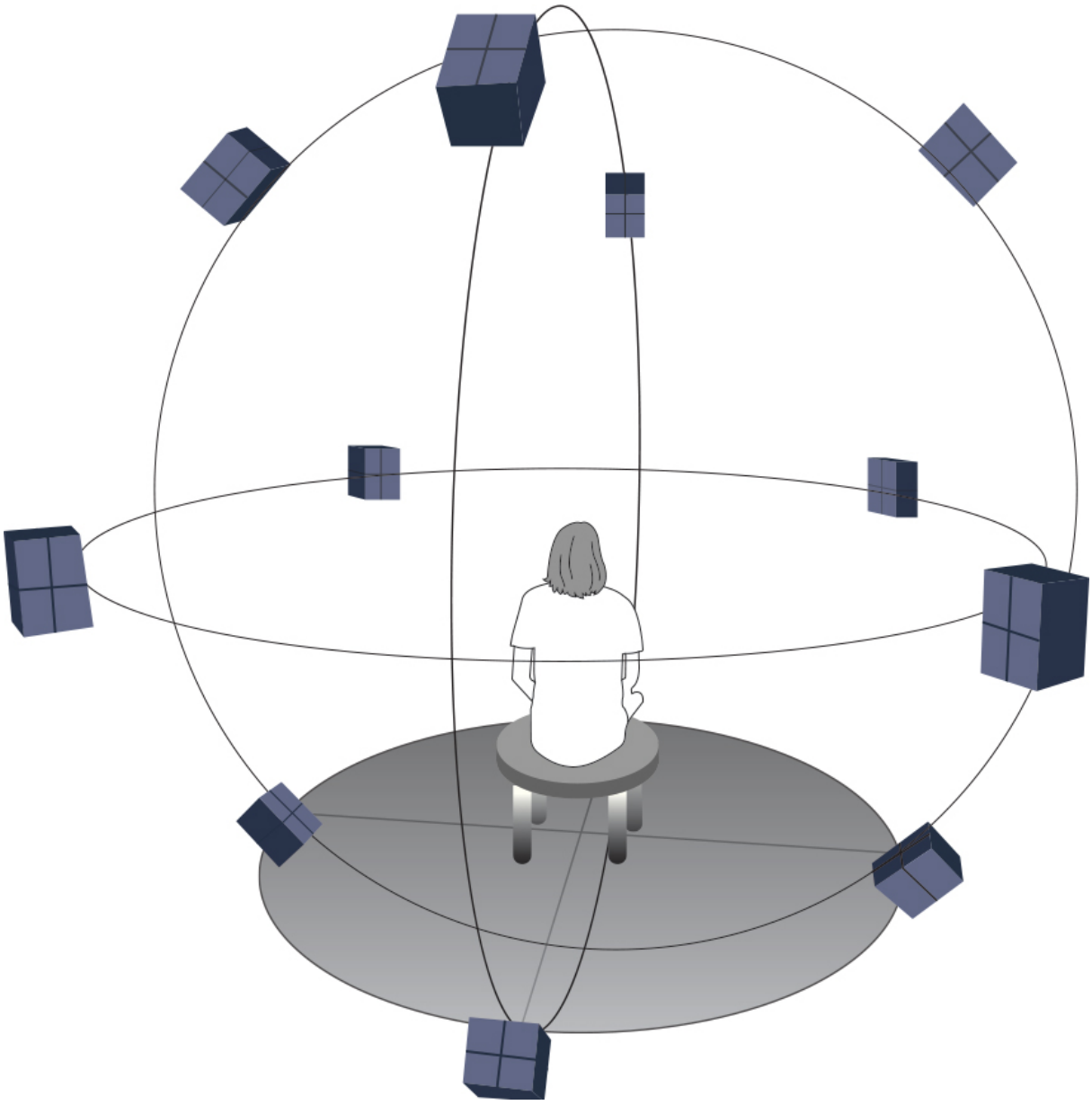
Ambisonics is an exciting alternative for multichannel sound recording that builds on the M/S stereo concept to deliver recordings that can capture all aspects of an environment as a fully 3D sound field. To do this, first-order Ambisonic microphones (we'll talk about higher orders shortly) have 4 cardioid mic capsules arranged in a tetrahedral array. The signal set created is referred to as A-format and this must be processed, via software or hardware, into a second signal set called B-format for use. B-format contains the spherical harmonics (a set of mathematical functions defined on a sphere, which are used to represent the sound field) and the component channels are identified by the letters W, X, Y and Z. These contain all the information required to reproduce the sound field including height, depth, elevation and distance. W is the sum of all 4 capsules and X, Y and Z represent 3 virtual bi-directional microphone patterns representing front/back, left/right and up/down. Conveniently, individual channels can be auditioned during playback. There are 2 file formats: FuMa and AmbiX. Here there is a difference between channel order; with FuMa, it's WXYZ where W has a -3dB gain versus the other channels. With AmbiX the channel order is WYZX with all channels having equal gain output.



RØDE NTSF-1 on location. Ambisonic recording enables you to capture 3D sound that can be decoded to work with a wide range of speaker setups.



First-order Ambisonics: mapping of B- format channels.



Idealised speaker arrangement for Ambisonic listening.

In order to play standardised B-format files they must be decoded. This is where Ambisonics starts to get really interesting; the same files can be rendered into mono, standard stereo, binaural stereo, quad, 5.1 and any other format the post-production software has the capacity to generate. This makes it an incredibly flexible starting point, enabling the field recordist to generate a single file that can be used to deliver any output format. And there's more good news: 'virtual microphones' can be created during postproduction

allowing the position and orientation of the microphones to be manipulated after recording. Sound sources can also be moved, panned, or removed if required. From a playback perspective, Ambisonics isn't a channel-dependant system, such as Dolby 7.1.4, and can work with any speaker array featuring a minimum of 4 speakers. In an ideal layout one would have a spherical array with the listener at the centre – most commonly the solution is a hemispherical layout with speakers above and below the listener's ears.

All this does rather sound too good to be true and indeed there are some issues to consider. To experience the full Ambisonic effect with speaker diffusion (not an issue if rendering files to stereo or binaural stereo for headphone listening), one needs to be in a relatively small 'sweet spot'. If the listener is moving about they're likely to experience some phase issues. And from a practical standpoint recording 4 channels (or more) of information can generate very large sound files. So far we've only been talking about first-order Ambisonics (FOA); higher-order Ambisonics (HOA) generate even more channels of information, leading to better accuracy in terms of spatial resolution – and even larger files.

Ambisonic Recording

The range of Ambisonic recording solutions is expanding but this is still a fairly small product arena. While a 4-input recorder is necessary for recording with a first-order Ambisonic mic, the entry level Zoom H3-VR (£180) follows the manufacturer's classic formula of offering an all-in-one low-cost solution. The 4 cardioid mic capsules are mounted atop a conical recorder base station. The screen is quite small but clear enough and can display all the necessary information. Unlike standard microphones, which can be held and moved, Ambisonic mics need to be placed in a fixed location and orientation is very important; any movement during recording will cause the entire sound image to shift. The H3-VR has a screw thread for tripod mounting and a built-in tilt indicator to

ensure optimal mic position. The basic orientation of the mic can also be specified, depending on what is optimal for your situation: upright, upside down or end-firing. The unit records Ambisonic A format or can be set up as a standard stereo recorder. Binaural monitoring for Ambisonic recording and playback is a useful feature for checking sound on location. Sample/bit rates up to 96k/24bit can also be selected.



Zoom H3-VR. A low-cost all-in-one Ambisonic recorder. Ambisonic microphones need to be set up in a fixed location to give image stability.

If you already own a Zoom H8 there is an Ambisonic mic capsule to add to your kit, the VRH-8. This is roughly the same price as the H3-VR, and features the mic capsules, built-in accelerometer and on-board Ambisonic decoder but takes advantage of the H8's recording capability. Once you have your recording, the free Zoom Ambisonic Player software allows you to process Ambisonic audio files and convert them to standard stereo, binaural stereo, custom

stereo (you can choose two points within the 360 degree spatial audio) and 5.1 surround audio files.

Other, somewhat intriguing all-in-one Ambisonic recording solutions, which evidence the ingenuity of a small manufacturer, are Brahma's Zoom Pocket ($750) and Zoom Studio ($1050). Both are based on Zoom H2 units – taking advantage of the unit's 4-track recording capability – but with capsule modifications to enable Ambisonic recording. The former looks indistinguishable from the basic H2n as the replacement capsules are mounted inside the existing grille. The latter has an additional 3D-printed enclosure featuring four larger diaphragm condensers. Both are supplied with microSD cards containing calibration files and Brahma's own software for converting A-format files to B-format. Brahma also makes an independent first-order Ambisonic mic, the Field 4 ($600), which is both compact and light.



MiniDSP ambiMIK-1.

USB first-order Ambisonic mics are a recent development and the MiniDSP ambiMIK-1 ($399) features its own preamp, analogue-to-digital converters and USB processor. It’s a very compact package that delivers recording capability up to 32bit/192kHz and, like many Ambisonic products, ships with its own suite of plug-ins.



Core Sound TetraMic.

Another stand-alone first-order Ambisonic mic is the Core Sound TetraMic ($625 direct from the manufacturer). Each mic is individually calibrated and comes with its own calibration files. The mic records 4 channels of audio in A format, and there is a custom version of the VVEncode plug-in (Mac and PC) that performs the A- to B-format (ambiX) transformation using the calibration data supplied with the TetraMic. The B-format file can then be decoded using any VST or AAX decoder plug-in. If you want to stay with the

VVAudio option, VVDecode ($25) is a first-order Ambisonic decoder that can produce speaker feeds for stereo, 5.1 or many other speaker configurations.



RØDE NT-SF1.

RØDE is part of the Freedman Electronics Group, which acquired the original SoundField company in 2016 – leading to the collaboration that developed the RØDE NT-SF1 (£1000). This is now

an established, broadcast-quality microphone for Ambisonic recording. RØDE also has the companion SoundField plug-in, a one-stop-shop for A- to B-format conversion and sound manipulation. It gives you the ability to change the polar pattern of virtual microphones in post-production, allowing the creation of shotgun-type patterns as well as omni, cardioid and figure-8 patterns. It also enables setup for all the common surround-sound formats, such as 5.1, 7.1.4 and custom formats and speaker arrays.



Sennheiser Ambio mic mounted in Rycote Cyclone windshield.

The Sennheiser Ambeo VR mic (£1300) is another high-quality solution. The 4 KE-14 mic capsules can be exposed during recording or remain securely covered by a conventional-looking grille. For A-format data conversion, Sennheiser has the AMBEO A-B converter plug-in (VST, AU and AAX format) as a free download. In terms of audiophile sound quality and extremely low self-noise, the Nevation VR (€3999) has a number of unique features including heated 33mm capsules to ensure parameter stability in different climate conditions

and a switchable LED light for self-illumination. SoundField is also still in the business of producing microphones designed for professional and broadcast applications. The DSPS200 (£3500) is the first SoundField microphone that doesn't require a dedicated microphone control unit. Like the other mics we've been looking at, the 4 capsules output to A-format and the company's SPS200 Surround Zone software can be used to decode to B-format and create the desired mono, stereo or surround formats.



Soundfield ST450 microphone.



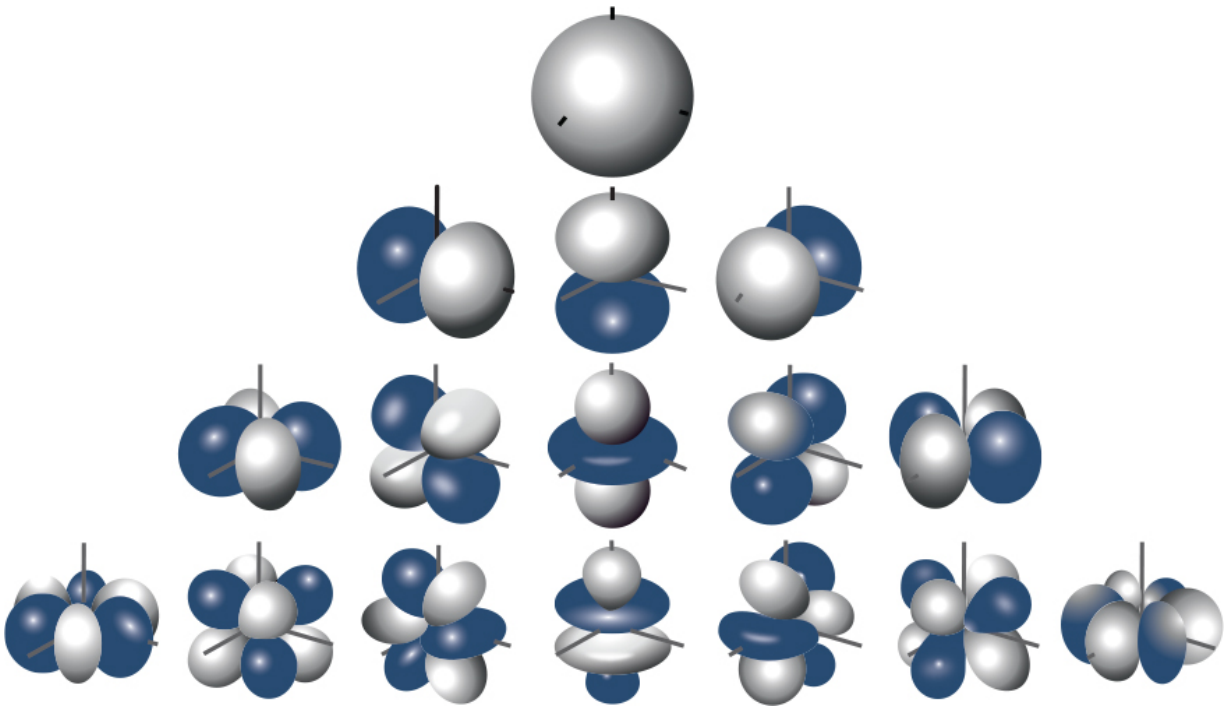
Soundfield ST450 controller unit.

The SoundField ST450 MKII Kit 1 (£7200) is a lightweight, battery-powered surround mic system for location recording comprising the mic and control unit. This serves a number of functions. It's a high-quality, low-noise preamp and as well as supplying phantom power, delivers power to the small heating element that keep the capsules condensation-free under normal operating conditions. It also outputs stereo left/right, M/S and 4 channels of SoundField B-Format. SoundField produces two additional surround packages – the DSF-2 MKII Microphone system (£12,000) and the DSF-B MKII (£20,000) are both combination mic/processor units, primarily designed to provide simultaneous digital surround and stereo recordings for sports stadiums and concert halls.

Higher-Order Ambisonics

All the mics we've looked at up to this point have been 4-capsule, first-order units and at the time of writing this is still the dominant format. However higher-order mics exist and this will be an expanding product arena in the next few years. Higher Order Ambisonic (HOA) recordings generate more channels of information. This leads to better accuracy in terms of spatial resolution and overall clarity. To do this, more microphone capsules are required, along with multichannel recorders or DAW/interface solutions with an

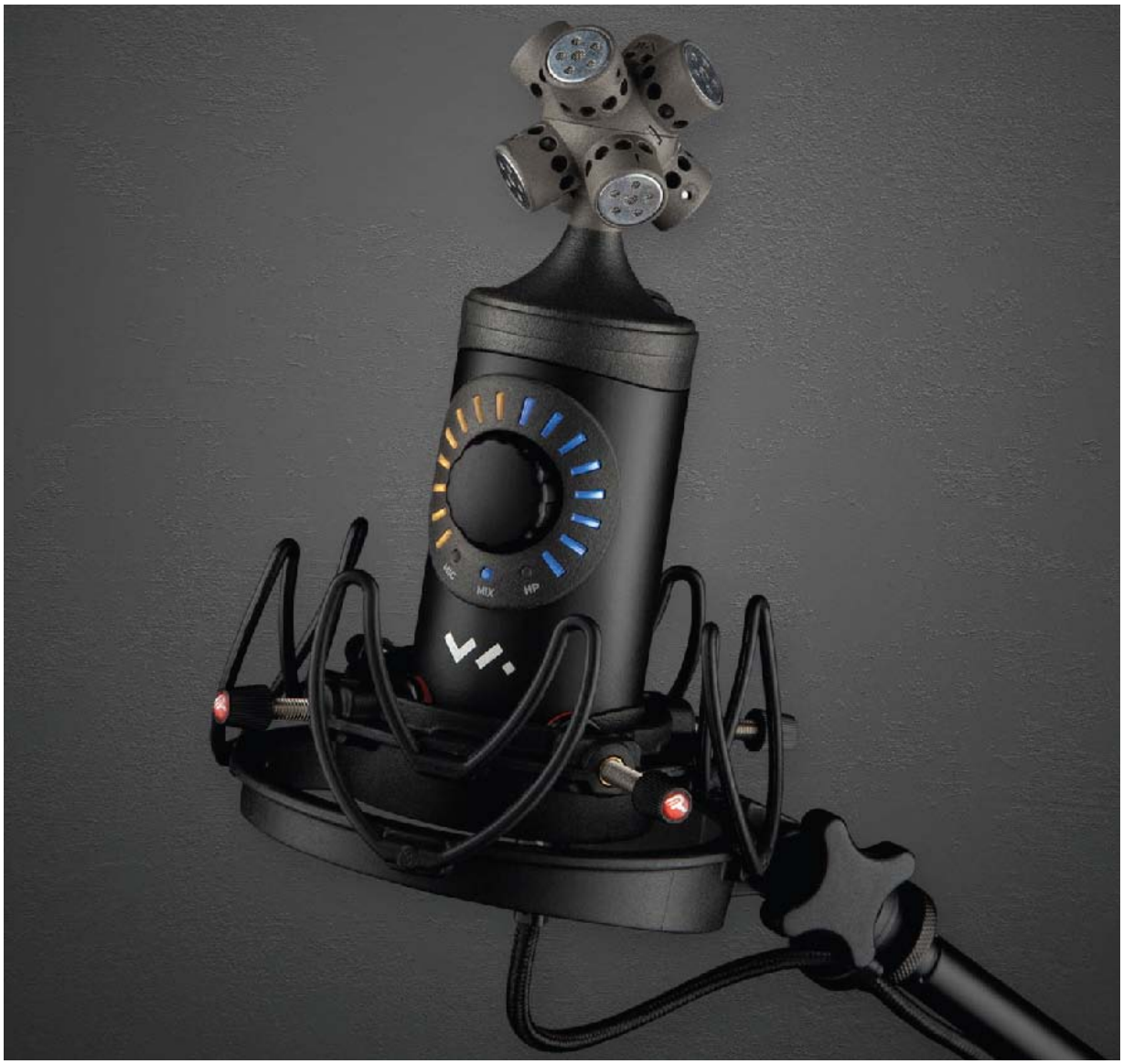
adequate track count. The Core Sound OctoMic™ ($900) is a second-order Ambisonic mic with 8 cardioid capsules. It maintains the same small footprint as the TetraMic and comes with a plug-in (VST and AAX) to perform A- to B-format conversion. The plug-in outputs standard 9-channel second-order B-format. The Voyage Audio Spatial Mic (£1100) represents a likely format for more products that will follow. As an 8-capsule second-order mic, it features both USB and ADAT connectivity and is USB powered. It's designed for both studio and Ambisonic field recording. The Voyage Audio plug-in allows you to set or crossfade between different polar patterns and virtual sound directions. The Brahma Field 8 ($1200) is another compact 8-capsule second-order Ambisonic microphone. The mic outputs to two 12pin DIN connectors, one carrying channels 1–4 and the other carrying channels 5–8. Cables and shock mount are included. Filtermatrix files to produce output signals as second-order AmbiX B format or SPS (Spatial PCM Sampling) are provided, and overall positional resolution is much higher than with a first-order microphone.



Spherical Harmonics up to degree 3, as used in third-order Ambisonics.



The Core Sound OctoMic™.



Voyage Audio Spatial Mic.



Brahma Studio 4 and 8. (Photo: Bhawesh Bhandari)



Zylia Pro ZM-1S.

Three issues facing the development of higher-order Ambisonic microphones along ‘traditional lines’ are size (due to the increased number of capsules), connectivity and the need for portable digital recorders with ever-higher track capability. This is leading to some technical innovations. For example, the Zylia Zm-1 ($1500), a third-order Ambisonic microphone featuring 19 omnidirectional digital MEMS microphones distributed over the surface of a sphere. The ZM-1 achieves 48kHz/24-bit recording over USB so you can record direct to a laptop or the specially designed ZR-1 field recorder (approximately $1400 – contact the manufacturer regarding ordering and price). There is also the Pro ZM-1S ($7400) microphone, which is the same mic but with a timecode input channel allowing for synchronisation of multiple ZM-1S microphones and other devices

such as cameras. Indeed, should you be in the market for multiple ZMS-1s for AV/VR applications there is the ZYLIA 6DoF VR/AR set (€32,600) which contains 5 ZM-1S microphones and ships with a raft of software plug-ins.



MH Acoustics em64 Eigenmike.

Another, quite extraordinary product is the MH Acoustics em64 Eigenmike ($15,000). This is a spherical array containing 64

omnidirectional electret microphones. The mic decomposes the sound field into spherical harmonics of up to sixth-order. It’s powered by, and connected to, a computer (Mac or PC) over Ethernet. Signal transmission from the capsules is achieved using Audinate’s Dante Virtual Soundcard software. The mic contains programmable gain preamplifiers, microphone boost and 24-bit A/D conversion. Microphone calibration is also achieved onboard. For control, processing and recording of its signals there is the EigenStudio® application and there’s a set of VST plug-ins.

In the field

Using surround setups and Ambisonic mics on location poses many of the same problems as more traditional mics, as well as having some specific requirements of their own. They still need to be protected from the elements and shielded from the wind. Here, the same basic rules apply and as the mics aren’t especially different in shape or size, some existing solutions from Zoom, RØDE, Rycote and MOVO can be adapted. Dedicated solutions like Rycote Cyclone Stereo Ambisonic Windshield Kit (£777) are available. Zylia also makes its own windshield ($77).



Zoom H3-VR on location.



Rycote Cyclone Stereo Ambisonic Windshield Kit. Wind protection is vital for Ambisonic mics and other multichannel setups.

As mentioned earlier, Ambisonic mics are not designed to be used in a roving fashion and tripod mounting is required. Also, placement and orientation in relation to the environment/events you want to capture is very important. Yes, there is lots of potential for sound image manipulation in post, but all of the issues we mentioned in Chapter 2 concerning surface reflections, echoes and overall ambiences still apply. Arranging a string quartet around a surround mic makes perfect sense – but recording a forest soundscape? In the absence of an obvious (central) location, monitoring in the field during setup becomes an even more important part of recording practice. Units such as the Zoom F8n and the Sound Devices Mix pre range offer A-format to B-format monitoring (as stereo) in real time.

Another important issue for all multi-mic arrays or multi-capsule mics is calibration; all 4 channels supplied by a first-order mic should be going to a recorder that can be properly calibrated – ideally to within a tenth of a dB. Input channels should be ganged together so that one knob controls all four (or more) mic preamps simultaneously. While contemporary digital recorders should have accurately matched preamps it's worth checking, especially with older units. This can be done by zeroing each pre, ganging them together and then using a hardware signal generator or phone app to send pink noise to channel one and lining it up to -12dBFS, and then sending the same signal to each input in turn. Once connected to the mic, the scale deflections will of course be different because the mic will be listening to the environment in full 360 degrees so the different capsules will be getting both on and off axis sound, reflections and so on.

Due to the nature of placement at a central point, an Ambisonic mic is effectively what, in studio parlance, we would call a 'room' mic – so the recording may not capture the detail of specific close sounds you require. If this is the case combining an Ambisonic mic with one or more 'spot' mics is the solution. This is especially important if you are planning to use the recordings in a VR environment where head tracking is being used. If you are going to do this, you will need a time-delay reference so the Ambisonic and close mic recordings can be correctly aligned in postproduction. A simple, audible click or clap is all that's required.

Multichannel Sound Recording Case Study #2: David Monacchi, *Fragments of Extinction*: Acoustic Biodiversity of Primary Rainforest Ecosystems

In 1998, while recording Italian natural soundscapes, Monacchi observed that the biophony of untouched forest ecosystems should exhibit structured behaviour, maximising efficiency within diversity. This led him to conclude that, if properly reproduced, soundscape recordings of these ecosystems could be a powerful means for raising awareness of acoustic biodiversity and the importance of a heritage being destroyed by rapid deforestation and climate change. In 2002, with the help of Greenpeace, he travelled to the equatorial Amazon to record in an undisturbed area of old-growth rainforest. His experience of extremely balanced acoustic systems produced by hundreds of species of insects, amphibians, birds and mammals vocalising within stunningly regular circadian cycles confirmed his initial hypothesis. Since then, he has been integrating scientific inquiry into the configuration of these soundscapes and ways in which their aesthetic features can be explored and rendered for the public.



David Monacchi on location in Borneo with em64 Eigenmike. (Photo: Alex D’Emilia)

Intensive fieldwork was required and Monacchi travelled to the world’s largest remaining areas of primary rainforest along the equator and recorded 24-hour sound portraits of various habitats. The work is especially poignant because the most recent International Union for Conservation of Nature (IUCN) projections indicate that half of the original species (the great majority of them unknown to science) will be extinct by the end of this century.

During these field trips Monacchi paid great attention to the recording process. As compared to previous scientific investigations, which have mostly focused on single species’ vocalisations and behaviour, his recording strategy adopted a broader ecological perspective, designed to capture as many components as possible of the complex soundscapes. Spatial

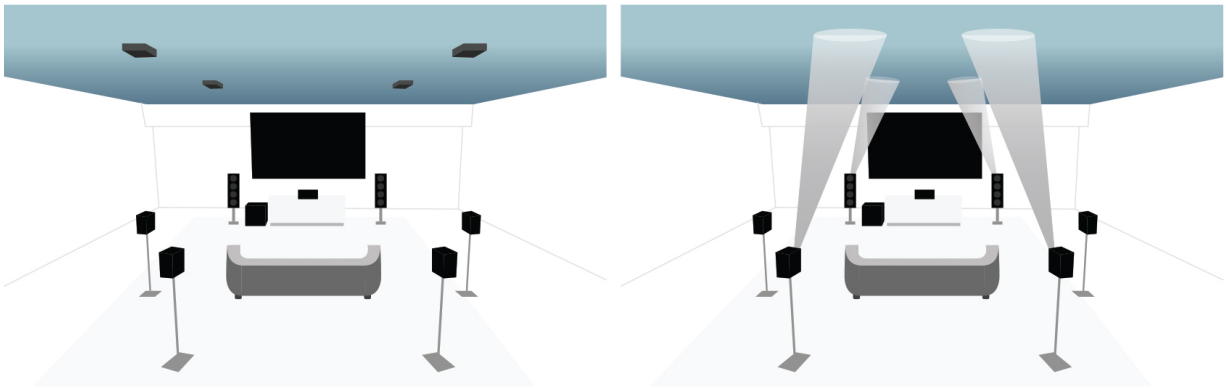
information of a given acoustic environment became a key element in understanding the complexity of its organisation and its subsequent presentation. In high canopy forests, sounds come from every direction, including above (such as birds and monkeys) and below (such as amphibians and insects) as well as the general proximity of the listening position. He employed ‘space-inclusive’ and ‘space-preservative’ standards and used experimental mic techniques to facilitate the accurate reproduction of these ecosystems over periphonic loudspeaker arrays. To achieve this objective, he developed a recording protocol with three simultaneous systems: quasi-binaural stereo (using a self-built experimental sphere mic with Schoeps CCM2S microphones); first-order Ambisonics (in collaboration and technically sponsored by SoundField for all ST-series tetrahedral microphones) and high-order Ambisonics (in collaboration and technically sponsored by MHAcoustics using the em32 and em64 Eigenmike). In addition, in specific situations, a self-built hyper-directional parabola microphone, and a pair of DPA4060 miniature omni microphones (self-worn for binaural soundwalks moving into the forest) were used.

The investigation of long temporal sections (over 24 hours of continuous recording) in such remote and dangerous habitats required the development of recording strategies for extreme conditions (humidity up to 99 per cent, sudden rainstorms, absence of electric power) and self-sufficient systems capable of adjusting the sonic perspective to on-ground, mid-floor and canopy species.

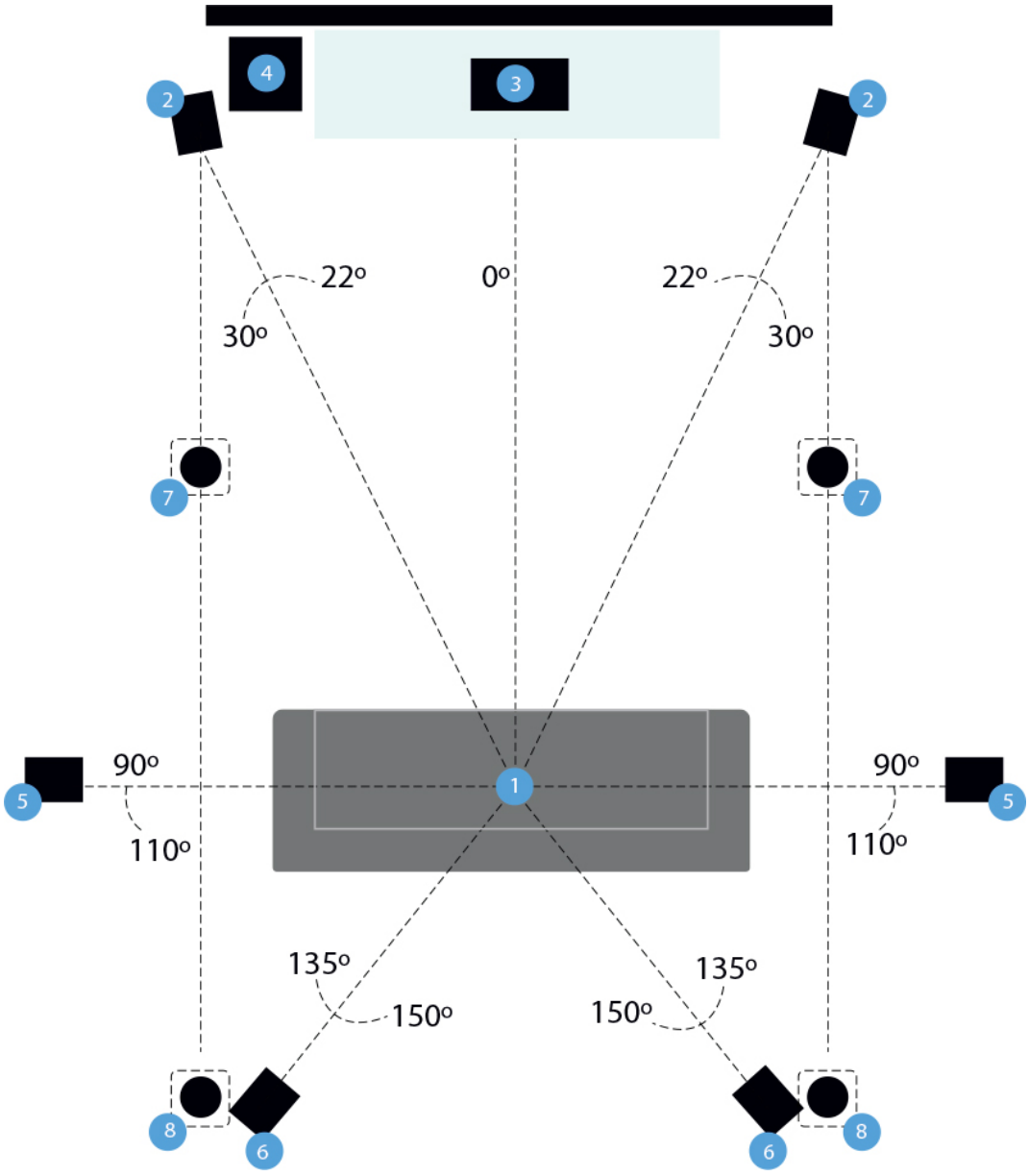
For further information about this ongoing project see:
https://www.fragmentsofextinction.org/

RECORDING FOR DOLBY ATMOS

The good news for phonographers is that recording for Dolby Atmos doesn't require new tools or techniques. Dolby Atmos began life as a surround technology for cinema installation in 2012, adding extra speakers for height channels to the more conventional surround setup. Atmos combines channel-based ambiences and sounds that are processed as 3D objects to create highly immersive audio experiences. Home cinema 5.1 or 7.1 systems can be adapted to become 5.1.4 or 7.1.4 with the .4 representing the additional overhead (or upward firing) speakers. Here the technology might have stayed if companies such as Apple Music, Tidal and Amazon Music hadn't started offering Atmos music mixes to support spatial audio streaming for the new generation of smart speakers, sound bars and earbuds. An Atmos mix is either an adaption of an existing recording or a multichannel composition environment. As a tool for the soundscape composer it offers a fascinating canvas to explore – I'll be looking at Atmos from this perspective in Chapter 6.



7.1.4 Dolby Atmos-enabled home cinema/audio setups with either ceiling-mounted or upward-firing speakers.



Dolby 7.1.4 speaker positions and angles for correct home cinema installation.



Smart speaker design showing 5-speaker layout and equipped with Dolby Atmos and spatial audio processing technology.

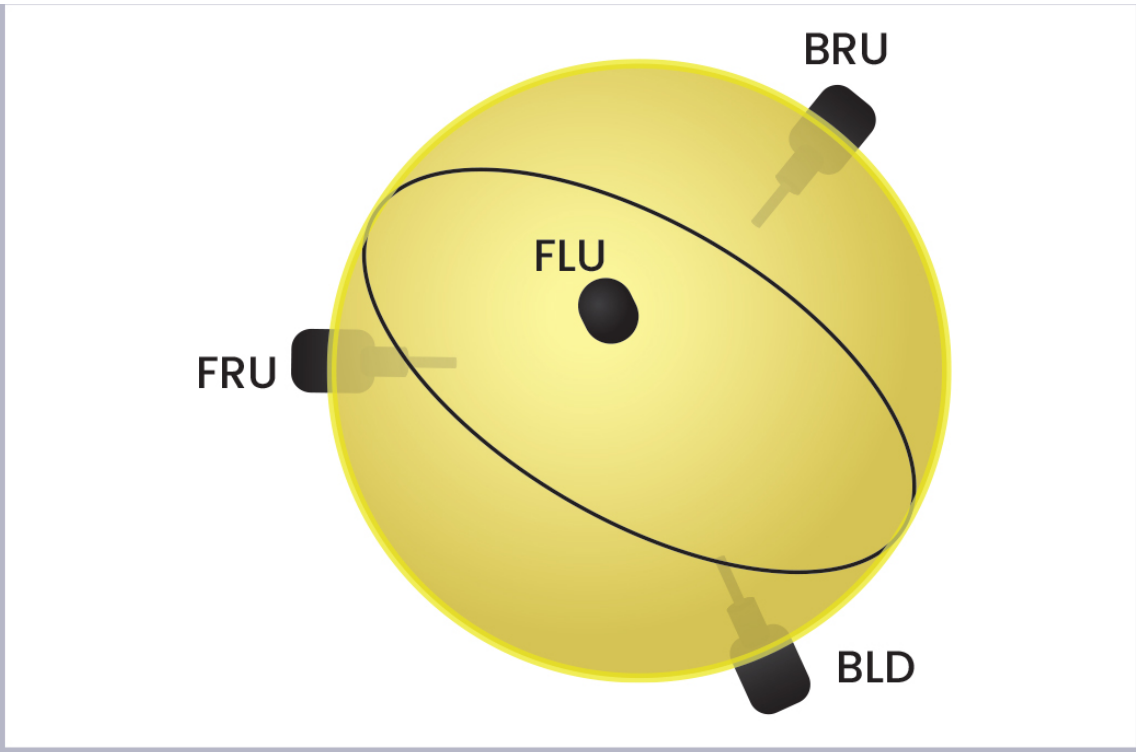
Multichannel sound recording case study #3: James Crutchfield, David Dunn and Alexandra Jurgens – underwater Ambisonics recording

Sound propagates very differently in water compared to air: it travels faster, dissipates less and different frequencies travel at different speeds. These factors make underwater sound difficult to analyse and interpret. While the idea of recording 3D underwater sound fields promises all the same benefits for sound localisation and specialisation as airborne sound, hydrophones have a fundamental limitation: they are omnidirectional transducers. Ambisonic recording requires the use of microphones with multiple directional (cardioid or super-cardioid) capsules. To overcome this limitation, in 2022 James Crutchfield began developing a unique hydrophone array which he named the hydroambiphone (HAP). This utilises 4 omnidirectional hydrophones mounted on the surface of a 12-inch diameter, 2mm-thick, hollow stainless-steel sphere. This is designed to deliver a first-order Ambisonic approximation, which is achieved because the acoustically opaque sphere creates an acoustic shadow that improves the directionality of the hydrophones.

In its first use, Crutchfield and collaborators David Dunn and Alexandra Jurgens began experimenting with the HAP in the field in the summer and autumn of 2023 during a 300-mile voyage through the Inside Passage of southeast Alaska, transiting from Juneau to Ketchikan for the purpose of monitoring humpback whales (*Megaptera novaeangliae*). The HAP was suspended using a 60ft steel cable and was deployed at depths ranging from 15in–50in. A Zoom F8n Pro was used to make 44.1kHz/24-bit recordings. The highly sensitive nature of the HAP created some issues, notably the capture of sound from surface waves, the boat's hull, sonar and engine and distant cruise liners. However, much of this could be removed

or attenuated during postproduction – specifically because of the benefits of Ambisonics recording (Crutchfield J. P., Dunn D. D., & Jurgens A. M. 2023). Since then Crutchfield has deployed the HAP to record humpback whale vocalisations in Maui, Hawaii, during their breeding season there, and in Monterey Bay, California.

Over 100 HAP recordings have been made to date and these led to several insights and discoveries about whale behaviours and vocalisations. Spatialising the social calls used by humpbacks during bubble-net feeding revealed previously unreported vocal coordination of different individuals demonstrating synchronisation and frequency shifting. Specifically, the HAP's spatialisation revealed that individual animals in bubble-net feeding groups were synchronously vocalising from different nearby locations.



HAP layout: FIG. 1: Hydroambiphone with hydrophone placements on the surface of a hollow sphere: (i) Front-Left-Up (FLU) (ii) Front-Right-Down (FRD), (iii) Back-Left-Down (BLD), and (iv) Back-Right-Up (BRU).

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5
PLANNING FIELD TRIPS



Simone Eringfeld on location in Antarctica.

WHAT IS A FIELD TRIP?

On the surface, this seems an easy question to answer; in the context of this book it's a visit to a location where we hope to make great-sounding field recordings. However, as we unpack the question further it becomes clear that it can mean many different things. To start with a basic field trip definition, let's propose 'a visit by an individual or group to a place away from their normal environment for the purpose of gaining first-hand knowledge and experience about the location, its attributes and features'.

Field trips may also be to places we know or have known, and here more intimate, personal knowledge will play an important role. Field trips are an experiential learning tool; delivering tacit knowledge that augments more traditional forms of learning. We will all have memories of a classic school field trip to a museum, factory, geological area or field studies centre. As field recordists our objectives are focused on documenting our chosen location through sound recording. However, the acquisition of knowledge and experience are still underlying motivations. And there may be a range of other agendas. Perhaps we are a researcher using sound recording to document communities, highlight environmental issues or capture sounds previously unheard; an artist wishing to use found sound in composition; a wildlife enthusiast keen to encounter and capture the sound of the species that fascinate them or an audio professional collecting sounds for use in relation to film or other media.



Elin Már Øyen Vister is a multi-disciplinary Norwegian artist, poet and composer based in the Røst archipelago at the far end of Lofoten/Lofuohtta. Recordings of puffins, kittiwakes and guillemots, the sounds of fisheries, and FM radio masts resonating in the wind paint a wild and intriguing sonic portrait of the location. (Photo: Alex Rishaug)

PRACTICAL CONSIDERATIONS

Preparations

Making preparations for a field trip can be relatively spontaneous – Jez riley French for example is a great exponent of the idea of simply walking out the door with your recorder and seeing what happens. For obvious reasons, a visit to a local wood will take far less research and planning than a trip to the Antarctic. However, having a clear sense of what needs to be achieved in advance increases the chances of a successful outcome ‘on the ground’. Location-based research may include geographical, social, political, ethnological, historical, ethical and environmental considerations, the identification of specific features (such as a river, church bells or unusual acoustics), migratory patterns and tide times. We also need to make decisions about the most appropriate recording equipment and the approach we will take, stowing/transporting kit and other provisions, making travel plans and practical arrangements such as arranging visas and safety precautions.



Getting ready for a day in the field – battery charging and kit lists. It's hard to over emphasise the need for thorough preparation.

Sometimes it will only be practical to make a single visit to a site, so objectives need to be clearly set and time managed appropriately. Sometimes a second visit following an initial assessment of the experience/environment will be highly beneficial. While the idea of an initial visit to watch, look and listen without recording may be a good idea in some circumstances, environmental conditions and soundscape content often change over time so it is possible things will be different when you return. Indeed, change itself may be an important aspect of the site you wish to document, so multiple field trips may be required. In this case documentation and file management becomes critically important and this needs to be planned for.



Good, thorough research makes planning field tips much easier; the more you know about the site, its fauna and flora, the better.

Long before any field trip takes place, I will usually start collecting research materials – collating online sources, books, journal articles, maps and newspaper content (if appropriate). On occasion, I may need to identify specialists or subject experts, make contact and conduct interviews. If I'm looking to explore different sonic characteristics over a range of locations I may need to identify multiple sites and define parameters for the investigation.

Some years ago, one of my colleagues, Professor John Levack Drever, came up with a list of questions (Levack Drever, 2002) a field recordist might like to ask themselves before going on a field trip. Many are thought provoking and considering them helps frame an approach. Inspired by his list, here are some questions of my own that I will work through during the research and planning phase of a project:

- What am I hoping to record?
- What perspective do I want to create? A static recording position? A mobile perspective that follows a path through the landscape?
- What is my primary subject? Ordinary, everyday sound? Unique sonic environments? Specific ‘rare’ sounds? Hidden sounds that need revealing through the use of special microphones and other devices?
- What’s my timeframe? Am I documenting durational aspects of the location or hoping to capture brief moments or fleeting occurrences?
- How important is embedded meaning to my purpose? Or am I purely looking for tantalising, inexplicable sounds?
- Do I want to interact with objects and features in the environment to reveal potential sources and latent sounds?
- Should I follow a strict plan as proscribed by a map or other scheme or should I be more open to chance encounters?
- What is my relationship to the location? Is it somewhere I’m very familiar with, have some limited knowledge of or is it somewhere I’ve never been before? How does this affect my choices in terms of what I record and when?
- Am I planning to use the environment as a backdrop for an in-situ voice over or narrative/commentary?
- Am I hoping to include or exclude human-made sound and utterance (clearly comprehensible speech, distant voices or general crowd noise)? Will there be other materials such as music or performances?
- What’s my position regarding ethics, privacy, copyright and permissions in relation to the material I record? Are the recordings purely for personal use? Will I be publishing them and if so, how? SoundCloud, YouTube and BandCamp or other?
- What equipment will I need to achieve my objectives?

Quite often, this process will lead to some form of document to collate my thoughts with sections covering key steps, a timeline, contact details, objectives and equipment lists.

Now, it’s certainly not the case that turning on a recorder always requires vast amounts of planning. This would exclude the possibility of spontaneously capturing something remarkable. Some of my favourite recordings are the ones I’ve made while on holiday when great opportunities have just presented themselves. However, being ready to take advantage of these situations required a degree of forethought; at the very least you need to pack a recorder.

Ethical Considerations

Field recording involves capturing sound in natural or human-made environments and this brings up several important ethical considerations. An awareness of these issues needs to inform the planning and execution of field trips and the ways in which we subsequently use recorded material.



Lucy Horswell, Stephanie Rogers, Nigel Ward, and Arthur Newton setting off to explore the Jurassic Coast at West Bay in Dorset. Most beaches in the UK are public spaces but not all. The Crown Estate, the Duchies of Cornwall and Lancaster, the National Trust, businesses and even private individuals own some. Always check the status of a location before planning a field trip in case access permissions are required.

Recording in natural environments must be done without disturbing wildlife or causing damage to the ecosystem. The presence of recording equipment or the person recording may unintentionally introduce noise or otherwise disrupt the environment. How do we plan for this? In some situations permissions will need to be arranged. Is an expert guide required? Should we be using a hide or other structure to conceal our presence? Can we get mics close in, or should we rely on shotgun or parabolic mics?

The nature of urban space is changing, and the rapid deployment of new surveillance technologies means that debates concerning what is public and private are of great contemporary relevance. Parks are generally considered public spaces but there has been a massive expansion of corporately owned leisure space. If you are recording in public or semi-public spaces – which includes markets,

shops and shopping centres – where voices and conversations might be recorded, it's important to consider whether those being recorded are aware and have given informed consent. Consent can be implied or explicit; even in public, people may not expect to be recorded.



Exploring the everyday: the author recording a rainy day in Leicester.

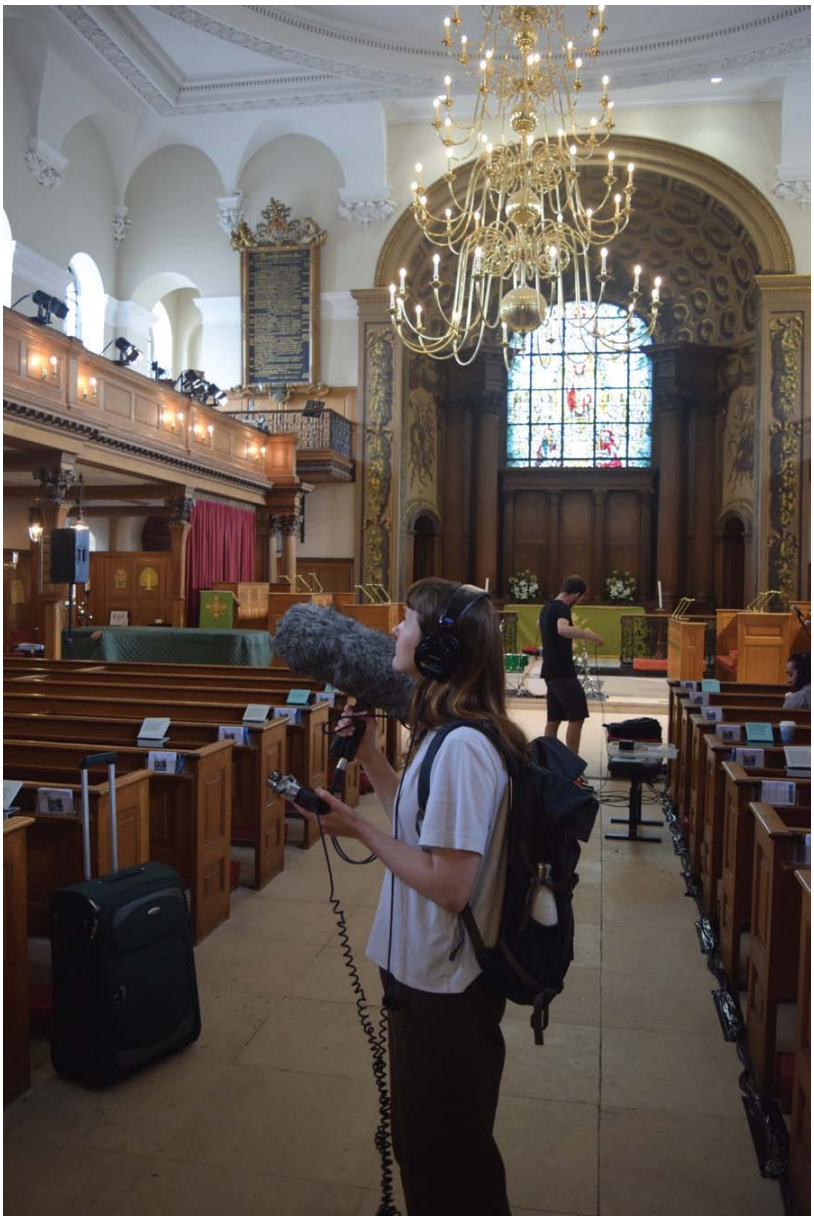
Ethical practice leans toward informing or asking permission when individuals can be identified. Transparency of intent is extremely important. Be clear about why you are recording and how the audio will be used, especially if the recordings will be published or used in media or research. Ideally one should be carrying consent forms and collecting names and signatures. Obviously, this isn't always going to be practical or possible and a degree of personal judgement comes into play here; there is a big difference between

undifferentiated crowd noise and a spontaneous interview with a passer-by.



Charity shop find – unexpected ultrasound. Iva Galovic and Lucy Wood discover a new sonic landscape. Shops are private spaces even though they are open to the public. If you plan to record, you need to seek permission.

In private space, recording without explicit permission is a clear violation of privacy and can be illegal. Some areas such as military bases, hospitals and schools may have legal bans on recording. There may also be issues of cultural sensitivity to consider, especially in relation to indigenous communities, sacred sites, rituals, music and spoken word content. In such circumstances permissions are essential and if recording is considered inappropriate, this needs to be respected. On a more general level that applies wherever you are recording, it's advisable to stop recording if someone asks you to so as to avoid confrontation or further ramifications.



Jo Pester exploring the acoustics of St Alfege Church, Greenwich. It's important to ask permission before recording in churches and other religious buildings.

Once we have made our field recordings, we also need to carefully consider how we will use them; editing that consciously misrepresents reality or changes the content of communications can be unethical, especially in journalistic or scientific contexts. In documentation it's important to acknowledge contributors. This can be people that directly assist with the recording process, but we also need to credit creative content producers such as street performers

– and some form of compensation may be required. This brings us to the subject of music copyright in relation to any material you may capture as part of a field recording. While this only really becomes an issue if you're planning to publish your recording in some form, there are some general guidelines. Traditional or folk music performed in public space is generally copyright free. However, seeking permission to use the recording and crediting the performers is the ethical thing to do. If an artist is performing their own material and retains copyright, permission needs to be negotiated. If your field recording captures someone playing a cover version of a contemporary song or an original artist track is playing on a radio, proceed to publish with extreme caution; the composer, lyricist or publisher still holds the copyright, even if the track is 'performed' informally in the field. Negotiating copyright clearance can be difficult and expensive and is best avoided.

EQUIPMENT PROVISION AND MANAGEMENT

It’s impossible to overemphasise the need to prep and check equipment in advance of a field trip. Specialist leads and adaptors, batteries, power packs and memory cards are all prime candidates to be accidentally left behind and I find it’s the absence of the small links in the chain that causes the biggest problems. Clear a surface, lay everything out, run through a checklist and test everything before putting it in the kit bag. When it comes to memory cards always pack more than you think you will need; SD cards rarely fail but it does happen. Also, if you are going to be recording a lot of the time it’s better to use a number of smaller memory cards rather than commit everything to a single large card – just a case of spreading the risk. Also, make sure any existing files have been transferred and that your cards have been freshly formatted. While old files won’t be written over, there’s no point in putting them at risk, and you don’t have a 32gig card if it’s already three-quarters full of data. Have safe, dry storage for cards that aren’t in use. I prefer to keep them on my person.



Equipment prep the night before the off: test everything, reformat memory cards, charge batteries and check everything off against a list before packing down.

Testing all kit in advance is extremely important – a lead with a bad, crackly connection can ruin an entire day's field recording. Start up recording devices and check settings. What exactly do you need to take? If you're going to have to carry equipment any distance, keep to a sensible minimum. Traditional mic cables are heavy and you don't need really long ones if the distance from your recorder to the mic is a couple of metres at the most. DPA make long (10m and 20m) ultra lightweight mic cables that are designed for use in the field. Both FEL MicBooster (UK) and Event Horizon & Services (US) will make custom-length cables using Mogami 2697 'miniature' cable and Neutrik plugs.

Hildegard Westerkamp began field recording in the late 1970s with a collaboration with Joan Henderson and Annie Holmes on a programme for Vancouver Co-operative Radio. Called *Soundwalking,* she would take a large Nakamichi cassette recorder and a pair of AKG mics on location and explore ambiences, close sounds and walking with the mics to create moving sound pictures. She explored natural and built environments, interior and exterior locations. As the work was for radio, she added comments and observations during recording to set them in context. As her recording practice developed this form of live annotation became an ongoing feature, giving her a voice that spoke from within the soundscape – drawing the listener into a nuanced and informed relationship with the environment as experienced from her perspective. *Kits Beach Soundwalk* (1989) is a good example of this style of work, which moves from documentary and exposition to mediation and composition. *Beneath the Forest Floor* (1992) is a composition based on recordings made in the Carmanah Valley and forests near Cowichan Lake on Vancouver Island, on Galiano Island and in Lighthouse Park. While also made for radio there is no running commentary. Instead, Westerkamp gives individual sounds the space for self-exposition and uses editing, juxtaposition and subtle musical elements to build up a sound picture that moves through a range of environments and moods.



Hildegard Westerkamp.

Kit Bags

Kit management is another thing to consider. Companies such as Zoom and Sound Devices make robust field cases and bags for their recorders. However, few companies now supply them when you purchase a recorder. Portabrace, Orca, Flyht Pro (Gorilla), Sachtler, Gator, Thomman, Gear4Music and DV247 all make kit bags of every conceivable size and shape. You can repurpose an old rucksack, but I find they are generally the worst solution in the field; to find anything you always seem to have to take everything out and things get lost in the deep, dark interior. If you don't have to walk far with your kit, a briefcase-style hard case with proper dividers that you can lay flat and open is far more practical. For urban field trips with a large number of students I use a medium-size suitcase on wheels; I can tow it along behind me and things are easy to locate. If I can possibly avoid it, I don't use a bag with a black interior because every bit of audio equipment seems to be black or dark grey and just disappears into the bag's lining. Mic pouches (ideally transparent) and small bags help keep everything in order. Investing in some Velcro cable ties is money well spent.



Finding the right kit bag makes field trips much easier. Harry Mcilroy recording a fountain with hydrophones.



When working with large groups of people, a good-size suitcase makes equipment access easy in the field if the terrain isn't too difficult.

If you plan to be on the move while recording with a hand-held device or microphone, you'll want to make sure your primary kit is all set up and that peripherals are ready to hand. Pockets are rarely adequate and a good recording bag or a camera-style waistcoat is simply more practical. If you're new to recording, practising at home with your 'rig' before a trip makes a lot of sense. Ergonomics is surprisingly important and you'll want to identify and resolve issues before any trip. For example, headphone cables are always a problem. Personally, I prefer a straight cable to a curly one; you can roll up the excess, cable tie it and slip it into a pocket out of the way.

FIELD TRIP ACTIVITIES

Let’s assume that your pre-trip planning has gone well and you’ve made it to your site with all the equipment you need and enough local knowledge to set about recording. If you’ve interrogated your planning questions in advance, you should (for the most part) know your agenda and have a plan in mind as to how to proceed. If you’ve never visited the site before some ideas will inevitably change when you’re on the ground – embrace spontaneity! Another important aspect of developing a professional approach to field recording is being able to make a discriminating judgement concerning location. Chris Watson points out that sometimes the environmental acoustics are simply wrong for what you’re hoping to achieve and it’s better to move on in search of a better location. Having the confidence in your own judgement leads to the best results and avoids a lot of time wasting.



Jo Pester recording the sound of musicians practising from a courtyard.



Robert Sookhan using a Soma Ether. Electric car charging points are a relatively new source of electromagnetic sound in the urban environment, with their own unique sonic signature.

When setting up equipment, take your time and be methodical; enthusiasm can take over, especially if you're a novice. Mistakes get made – like leaving the recorder on pause rather than record for half an hour. When moving from one location to another, be especially vigilant about not leaving equipment behind; it's easily done. It's the small items like adaptors, clips, wind jammers, mic pouches,

batteries, lens caps, gloves, hats, sunglasses and wallets that get put down and not picked up. I've trained myself to do one last visual inspection of the ground before moving on and it's paid off on a number of occasions. Don't just shove everything loose into your bag as you progress; further rummaging will just leave you with a mass of tangled cables. You don't need to be obsessive, just keep things as neat as practical and do a final kit check before you leave the field and a proper pack down when you get back to base. And don't forget to recharge batteries overnight if your field trip continues the next day.

Sound File Management

In terms of file management, it's a good idea to state where you are when you start recording so as to embed this information in the file. If you're moving to a number of different locations, add a verbal note each time you set up. This will make postproduction editing a lot simpler. It's also good practice to state what equipment you are using when you change anything significant. A good ear can identify the difference between a Zoom recording, a stereo shotgun and binaural mics, but make things as easy as possible for yourself. You may want to add other information to a verbal note, a scene description or note about weather conditions for example; it's surprising how quickly specifics fade in memory. All this can be edited out later. While it's possible to rename files in the field, I find this far too time consuming. File numbers however are easy to log and there is no reason not to log information and comments in a notebook if you prefer this to a verbal commentary.



Alan Cook using LOM Elektrouši to hear the hum from underground cables.

Field Trip Case Study #1: Annea Lockwood

Annea Lockwood started to explore field recording in the mid-1960s. Her experience of rivers as a child growing up in New Zealand led to a lifelong fascination, and was the inspiration for her archive of river sounds from all over the world. From around 1973 she started to create installations, first in New York, featuring the river recordings that she and a range of collaborators (such as Pauline Oliveros and Carolee Schneerman) had made. From 1981 she started exploring rivers from source to mouth, engaging with every aspect of environmental sound and recording stories and interviews with people who had lived experience of different aspects – such as the river pilots, fishermen, farmers and other residents. In her first sound map installation – of the Hudson river (1982) – interviews were presented on headphones with the river's sounds through speakers. Having come to see the people living on and around the river as an integral part of its ecosystem, like the fish or the vegetation along its banks, Lockwood mixed their voices in with all the other sounds for *Sound Map of the Danube* (2004) and also started incorporating hydrophone recordings for the first time. Both these sound maps are still presented as installations and there are CD releases: *A Sound Map of the Hudson River* (1989) and *Sound Map of the Danube* (2008).



Annea Lockwood recording at Chief Joseph Dam, Washington on the Columbia River. (Photo: Nate Wooley)

In 2024 Lockwood travelled up the Columbia River from its mouth near Astoria, Oregon to the Columbia Icefields of British Columbia. Travelling with the composer and trumpet virtuoso, Nate Wooley, they made field recordings along the way. Some were made at small parks and community boat launches, while others were recorded above and below the many small-to-massive dams controlling the river, so it was a challenging trip. The equipment they used was a Zoom H4 pro recorder, a Sound Devices MixPre-6, Aquarian hydrophone, Shure stereo mic and a RØDE shotgun mic. Other indispensable items

included a RØDE boom pole, Sony headphones, duct tape and at least one old towel.

The completed project includes the *Sound Map of the Columbia River*, along with a performance-oriented work that proposes an acoustic extension of the field recordings for the percussion and piano quartet Yarn/Wire and Nate Wooley's trumpet. Nate, a former editor of the journal *Sound American,* has written a text that acts as a diary of the trip alongside the social and political context of life on the Columbia.

Field Notes

Even if you are adding spoken word annotations to the start of recordings, a field notebook is usually an essential piece of equipment for any trip. In the field I like to take a moment to write down general observations about the environment – about topography, fauna, flora, personal feelings and conditions on the ground. I find re-reading these ‘in the moment’ commentaries at a later date combines with listening to create a much more visceral experience than can be achieved with sound alone. I’m not much of a sketch artist but I still find this useful at times. And I always take photographs as an aid to memory. Phone camera snapshots are perfectly adequate for documentation unless your purpose demands higher-quality images.



Binaural recording at Greenwich Market with Ben Koppelman.

Self Care

Don’t forget to look after the most important piece of field recording equipment; yourself! By all means buy takeaways en route or build in time for a pub lunch – personally I always pack food, snacks and plenty of water in advance and carry it with me. When I’m recording I’ve a very one-track mind and if distractions like eating or drinking

are too time consuming, I'll forget or delay with the result that it's easy to get completely strung out by the end of the day. This serves no good purpose and feels amateurish to me. A similar thing goes when it comes to dressing appropriately for the weather. Field recording is a practice that often requires a lot of sitting or standing still. Freezing to death or forgetting the sunblock will only make your experience less pleasant. Again, this is part of a professional approach and requires a degree of preplanning.



Make sure to dress for the weather – it makes field trips less of an endurance test. Lucy Wood on a blustery day in Whitstable.

Backups

At the end of the day when you're back at base, the field trip isn't really over until you've done the basic homework; transferring the digital files to a laptop and backing them up – preferably to the cloud or at least to external media. Check a few files at random to ensure

they contain what you expect them to and that the quality is OK, then either archive the cards as a further backup or reformat them.

Field Trip Case Study #2: Field recording in Antarctica – Simone Eringfeld

Between October 2023 and January 2024, Simone Eringfeld spent three months doing fieldwork aboard expedition ships in Antarctica as part of her PhD research. The project explores how sound impacts tourists' perceptions of polar environments. Central to her research were the field recording workshops she organised for tourists. Eringfeld provided participants with Zoom H1N recorders, JRF hydrophones and headphones, and guided them through an exploration of Antarctica's diverse soundscapes, capturing everything from penguin cacophonies to the sound of creaking ice.



Simone Eringfeld locating and capturing Antarctic ‘silence’. Antarctica is nicknamed ‘the silent continent’ – although its coastal regions are anything but.

Eringfeld also explored her own recording practice. She was particularly interested in locating and capturing ‘silence’, since Antarctica is nicknamed ‘the silent continent’ – although its coastal regions are anything but. Her work with the tourists meant that she did not have a lot of time for setup and monitoring. Instead, quick decisions about locations and subjects were required. The technical solution was a tripod featuring a Zoom F3 with 32-bit float technology, which allowed her to capture a wide range of sounds, including quiet ones, without the need for gain adjustments. The mics used were

MicBooster Primo EM272Z1 omnidirectional microphones for capturing soundscapes in stereo and a Sennheiser MKE 600 shotgun was used to isolate distant sounds such as whales or glaciers calving. An Ambient ASF-2 hydrophone was used to collect underwater sound. Icebergs were particularly fascinating to record: listening through hydrophones, some sounded like underwater rainforests or running rivers, while others resembled dripping taps, corn popping or fizzy ice lollies.

While the cold was less of an issue than expected, with temperatures typically between -7 and +5ºC, the strong winds and frequent saltwater sprays presented significant challenges. High-quality wind protection and waterproof covers, along with plenty of battery power, proved absolutely essential.

The resulting recordings have been analysed and written about in her PhD thesis. Additionally, some of the unique material is showcased in her podcast on Antarctic soundscapes. For more information on this project, visit www.simoneeringfeld.com

OceanofPDF.com

6
STUDIO PRACTICE

In this chapter I'm going to focus on the basics of sound file management, archiving and the hardware and the software tools you'll need to explore various types of documentary and compositional practice with field recordings. I'm not going to explore the specifics of different compositional approaches or the use of individual plug-ins, as this is outside the scope of this book. Likewise I won't be going into the same amount of detail concerning available products as I did with field recorders and microphones; for this I would need a companion volume. However, field recording is now firmly embedded in the digital domain so a basic level of understanding of studio tools and processes is necessary and more detailed knowledge will enhance your enjoyment of the sounds you record and propose new ways to explore their potential.



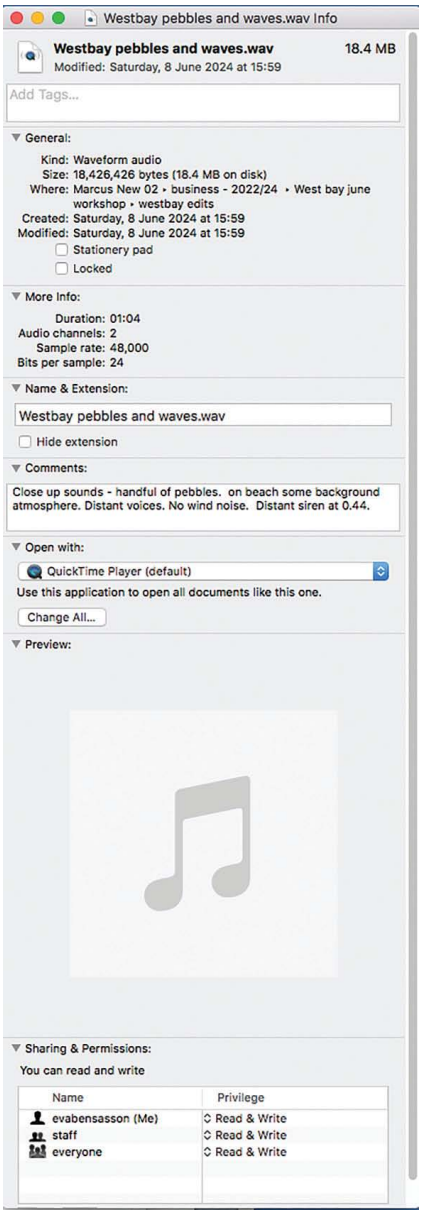
Home studio workstation.

It wasn’t until the mid-to-late 2000s that most people outside the world of professional audio could consider the option of a purely digital file-based workflow. Laptop and desktop computers and external storage media costs were starting to fall and companies such as Zoom, Sony and Fostex were introducing file-based recorders aimed at the semi-professional and consumer markets. Prior to this, most enthusiasts were working with moving-media digital formats such as mini disc or digital audio tape (DAT). Today, a

file-based process is the norm: simply a question of taking the SD card out of a recorder, putting it in a card reader (or connecting the recorder to the computer via USB) and transferring the files using a familiar drag and drop operation. From here it's a simple process to bring sound files into editing software. However, prior to this there is a process of file management, which needs to be more or less developed depending on your process and objectives.

File Management

If you're simply going to import a file, edit a selection and then name and export it for personal listening, raw data may not be that important to you. However, if you plan to create an accessible 'working' archive, both raw and processed audio will need to be named, classified and stored in some form that makes it easy to access. File numbers as assigned by a recorder are of little use; they will iterate every time you change or reformat a card. While date and time information will be different (assuming you've set up your recorder properly) this is of limited value. You can rename files using the recorder's interface but this is time consuming and usually has a character limit. A minimum requirement for good practice would suggest transferring files to a computer, naming them something representative of content, putting them into named, dated project folders and creating sub folders to differentiate between, for example, locations and recording techniques. The goal here is to develop your own filing and naming convention and stick to it. While some level of description in a file name is good, there's only so much information that can usefully appear. If you want to add depth to a description, you'll need to add metadata to the files.

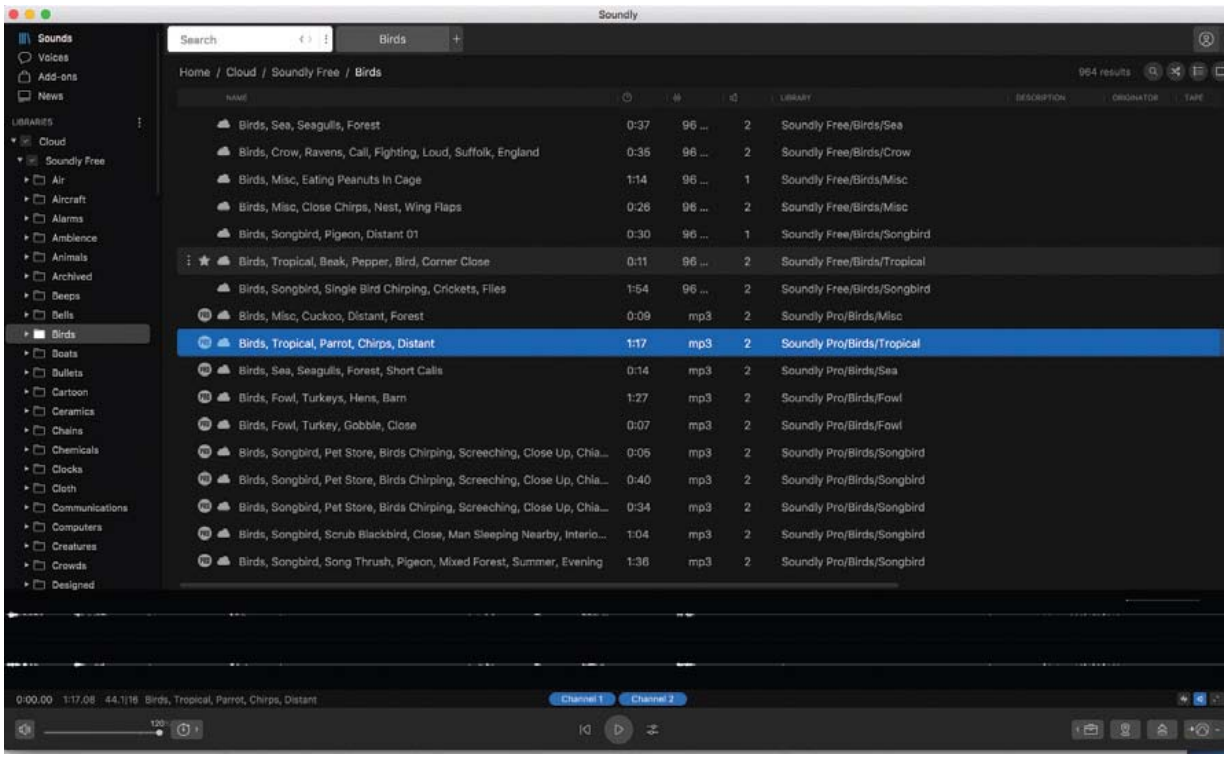


Sound file metadata window (Mac). Make metadata your friend; this way you can store useful information about sound content, location and so on, and it stays with the file.

Metadata is data about data: structured text that helps you to sort and identify file content. It makes specific files easier to find, use and reuse. Basic metadata includes an author's name, date created, date modified and file size. Metadata stays with the file so it is also a way of identifying ownership for copyright purposes. Anyone planning to share recordings is well advised to develop a good relationship with metadata. Metadata can be written or edited from

the file menu. For example, on a Mac, highlight a file and press [command] and [i] to reveal the metadata pane. Here you can add information in the Comments section. Make sure to specify ‘read only’ for anyone other than yourself.

In the short term (or if you’re not a prolific producer), plenty of well-named folders, sub folders and named files with good metadata content will serve you well. An Excel sheet is a relatively simple way to bring all the information to a central reference point. Beyond this, you may want to consider some form of archive/librarian software. For Mac users, iTunes can be used effectively but basic settings need to be modified. It’s very important to specify WAV or AIFF in Import Settings in Preference, otherwise your hi-rez audio will be converted automatically to MP3. Metadata can be added using the Artist Info menu. Other library software you might like to consider includes Soundly (free and paid-for plans with various levels of sound effects cloud library access), Soundminer ($199–$899) and BaseHead ($249–$549). All have Mac OS and Windows versions. In addition, Mac users may like to explore Iced Audio’s AudioFinder ($69) or Monkey Tools’ Library Monkey ($129–$499). Windows users have Liqube Audio Resonic Pro ($69) as an option.



Soundly: organise your own files and access cloud-based SFX libraries.



Universal Category System (UCS): extremely useful if you're planning to share files; most large databases can now work with the categories.

So far, we've talked about filing and retrieval on the assumption that you're not planning to share content more widely. If you are, you should consider using the Universal Category System (UCS). This was developed as a means to standardise the categorisation of sound effects for film, television and other media. It's also a filename structure and there's a set of tools that use the category list (753 categories and subcategories) to help organise sound effects libraries for creators and users. Most large databases can now work with UCS.

STUDIO HARDWARE

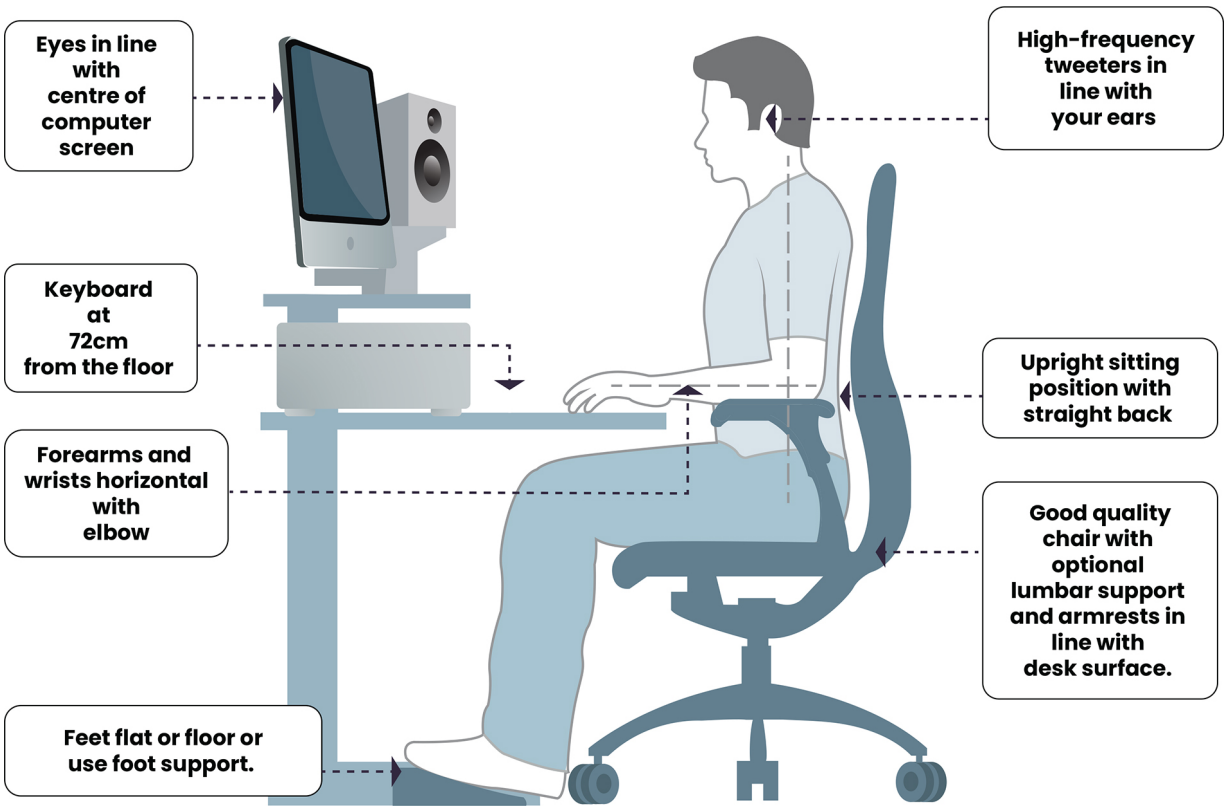
Computers and Workspaces

Both Apple and Windows machines are equally capable when it comes to editing sound. Audio is nowhere near as demanding as video and even a 10-year-old computer will generally be up to the task, even if things do take a little longer. Having external hard drives for running audio sessions hasn't been necessary for a long time, however USB external drives (or cloud storage) remain essential for backing up your work.



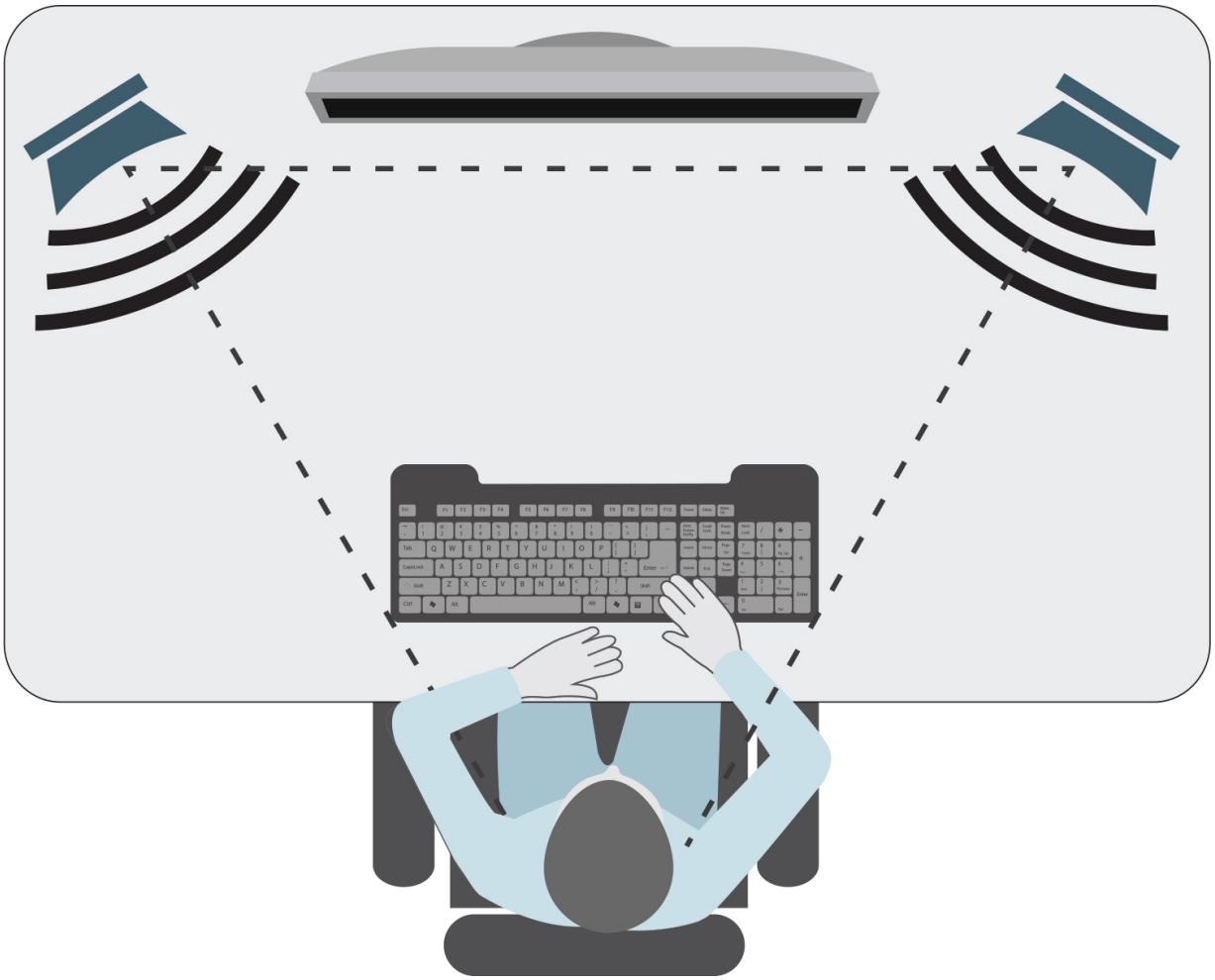
Studio workstation with Adam A7V monitors and RME Fireface UCX interface. In this setup the sound monitors are at a good height, but the screen is a little too high; seated, your eyes should be in line with the centre of the screen.

It’s also important to consider your working environment; long-term computer use can lead to a range of repetitive stress injuries and laptops are particularly problematic, rapidly causing shoulder, back and neck pain. I recommend either using a desktop machine or creating a desktop-style workstation using a separate screen, keyboard and mouse (or other device). The laptop’s screen can then function as second monitor, and you can never have too much screen real estate, especially if you are using lots of plug-ins. Being able to move the laptop away from your working position is also good as it distances any fan noise. Ideally your eyes should be in line with the centre of the main screen and your lower arms level from the elbow to the keyboard. A properly supportive office-style chair is much better than one repurposed from the kitchen. Aim to keep your feet flat on the floor – unless you are using a kneeling chair.



Computer workstation optimised for working with audio.

Use either tabletop or floor stands for your sound monitors. Height wise the tweeters should be roughly in line with your ears because high-frequency sound is highly directional. Don’t place speakers right against a wall, as this will exaggerate their bass response; 10 –15cm is about right. At the same time, don’t have them more than 1m from a wall as this may lead to an uneven bass response caused by reflections. Try to avoid having them within 1m of a sidewall or in a corner. If there is no option, some monitors offer frequency response adjustment to compensate. If you can set up your workstation equidistant from the sidewalls with your listening position at approximately 35–40 per cent of the distance between the front and back walls, this will give you the most balanced response.



The distance between the monitors and your ears should be the same in order to create a listening ‘sweet spot’.

How far apart should the monitors be? Opinions vary, but for small monitors like Genelec 8010s, 1m is considered adequate. For anything larger, 1.5–1.7m is the figure to aim for. Whatever the distance between the speakers you should be sitting with your ears roughly the same distance from the centre of each speaker, so as to create an equilateral triangle that places you at the listening ‘sweet spot’. Angle the speakers in so they are aiming at your ears – you can adjust the exact positioning to taste. The computer screen should sit as far back as practical to leave free space for the sound to defuse into the area in front of you. If you can get your workstation setup correct, remarkable things happen; the sound will appear to inhabit this central space rather than sit on the speaker cones.

If you want to explore studio acoustics in more detail all spaces have resonance frequencies, which are referred to as room modes: standing waves created as sound moves between sidewalls, floor and ceiling. This causes a unique pattern of frequency distortion, reinforcement and cancellation. Room modes can be analysed using software (such as XTZ Room Analyser 2 Pro or REW) and some form of wall and ceiling treatment (foam or covered Rockwool panels) and corner bass traps will be needed to properly manage the acoustic environment.

Headphones

Good headphones are very important, especially if they are going to be your primary working and listening tool. Standard ear buds aren't ideal; they're designed to make music sound good so often overcompensate by boosting the low end. Long-term wear is also extremely fatiguing. Closed-backed studio-style phones of the type you might also use on location are fine but are generally not that comfortable in the long term as they clamp tight to the head in order to isolate the listener from the surrounding environment. Low frequencies tend to sound over pronounced, and the closed cup design can lead to unnatural resonances. Also, there is nowhere for heat build-up to go so they can become sweaty and uncomfortable. Open-back headphones sound much more natural and can be worn comfortably for extended periods of time. They are a bit more expensive, but this has always felt like money well spent to me, especially as they will also vastly improve your general experience of mixing music if that's also your thing. And indeed, every aspect of computer audio will sound better, so you can justify them as a leisure expense as well! Options include Philips SHP9500 (£85), Beyer Dynamic DT990 (£120), AKG K702 (£150) or K712 (£280) and Sennheiser HD600 (£265).



AKG K712 – open-backed headphones are preferable for studio mixing and editing; they are more comfortable to wear because heat can dissipate and they sound more natural than closed-back models.

Audio Interface

You can connect a computer to an existing hi-fi system or dedicated sound monitors by using the computer's headphone output and a standard mini jack 3.5mm cable with either RCA or XLR connectors and there are Bluetooth speakers as well. While these options will work, and sound passable, it's far from the ideal solution; the headphone output isn't optimised for sound quality. Even the lowest budget audio interface will give better results and you will also get a dedicated headphone amp with more control over level – usually a dedicated rotary control. Most modern interfaces connect to the computer using a standard USB cable. Many are class compliant which means you don't need to install driver software, so they are

basically plug-and-play devices. As well as a headphone amp, an interface gives you dedicated balanced outputs (on TRS jacks or XLR connectors) to your sound monitors and will have one or more microphone and line-level inputs.



Focusrite Scarlett Solo – you don’t need a lot of inputs and outputs if you’re working mainly with sound files of field recordings.

Interfaces come in many shapes and sizes, with more or less inputs and outputs. There are interfaces for all budgets – generally speaking, the more expensive units give you better build quality, higher-fidelity sound, more input/output channels or all three. For most people working with field recordings, a 2-in/2-out interface will be suitable. This will enable you to record in stereo in the home/studio environment if you need to and, as many are bus powered, it will also give you the option of using a computer/interface for high-quality recording in the field. Well-known and reliable brands

include Behringer, Sub Zero, M-Audio, Presonus, Focusrite, Arturia, Mackie, Roland, Universal Audio, MOTU, Audient, Lewitt, RME and SSL.

Monitors

Hi-fi system stereo speakers are designed to make music sound good – they have a tonally coloured profile – boosting certain frequencies to get the best results. This is great if it makes the music listening experience more enjoyable, but totally not what you want if you're working with field recordings to make critical adjustments. Hi-fi speakers are also designed to sound good from anywhere in a room, from any direction and deliver constant results when you move about. What we need is spatial accuracy and detail for close listening. Studio monitors are designed to deliver a flat frequency response for near-field listening. When you sit in front of them in the 'sweet spot' you're hearing the recording warts and all.

So, if the level between the left and right channels is slightly unbalanced, you can adjust to compensate. Too much low-end rumble? As it's clearly audible you can remove or minimise it. Planning to do some complex lay-ups or sonically-invasive processing? Then you'll hear all the crucial details. Also, the majority of hi-fi speakers are passive and rely on an external amplifier to drive them. They have internal, passive crossover networks, which are designed to distribute the frequencies to the woofer and tweeter. Poorly designed, budget crossovers can introduce distortion and compromise overall sound quality. Most contemporary studio monitors are active units and have one or more built-in amplifiers; when the drivers are individually powered, sound reproduction is much more precise.

Thankfully there is a lot of competition in the studio monitor market and this has driven prices down. You can now buy a pair of functional monitors for as little as £80 that will definitely be better than hi-fi speakers. At the same time, nowhere is the old adage 'you get what you pay for' more true than in relation to the sound quality

of studio monitors. Lower-cost monitors tend to have quite low power amps and are more likely to evidence some self-noise than more expensive, higher-powered options. Poorly designed enclosures built from MDF or plastic can have resonance issues. More expensive units have solid aluminium enclosures and the combined quality of amplification and driver design will deliver superb, natural-sounding results. They will also have controls to calibrate the monitor for your listening room and speaker placement.



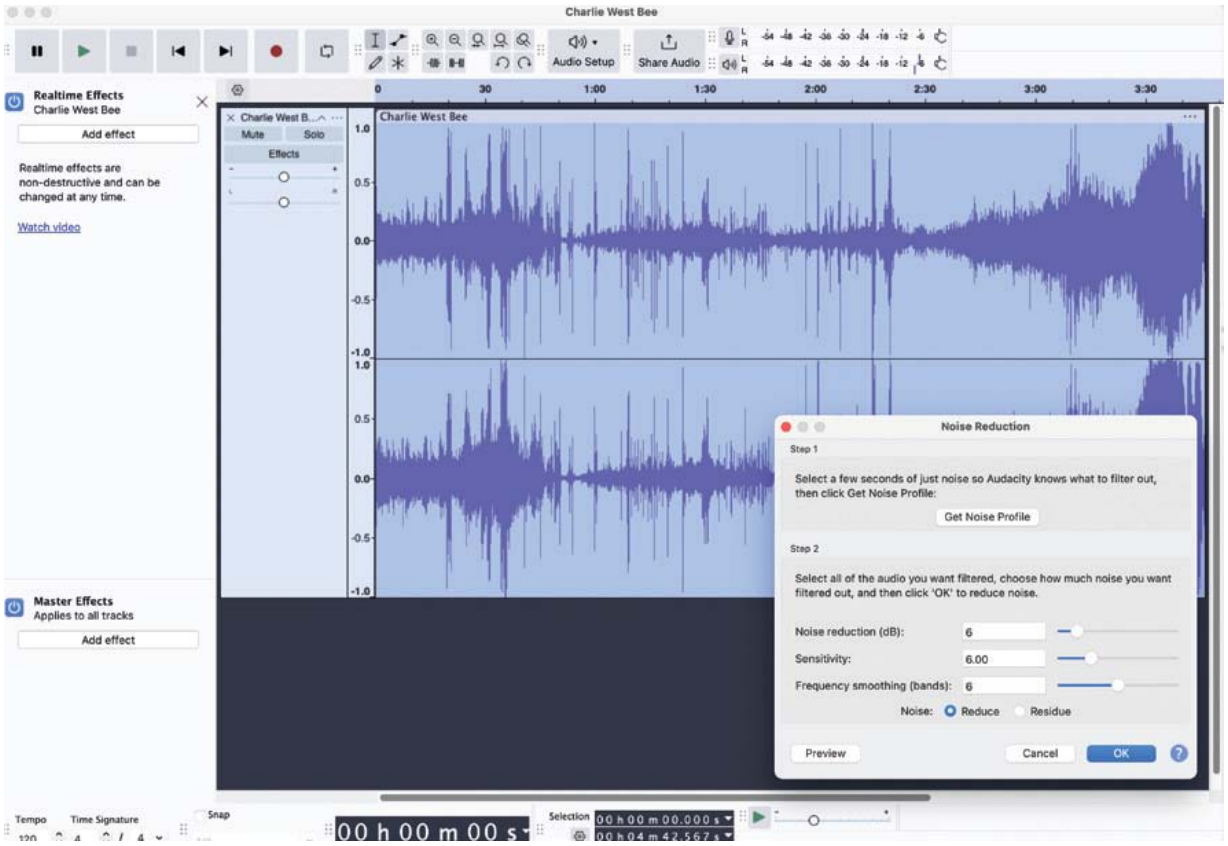
Home workstation featuring Neumann KH120 monitors. While an external screen is essential if you plan to work long hours, you don’t need a brand new high-resolution model if you’re just working with audio. This old Apple Cinema Display still does the job just fine.

Field recordists don’t need the larger, higher-powered units designed for music mixing rooms. A power rating between 40W–100W will give you plenty of headroom to run the monitors at low volume with zero chance of speaker distortion, low self-noise and

pristine sound quality. Models with 3in–5in woofers are good, and don’t take up too much physical space. It is possible to spend thousands of pounds on near field monitors but realistically a budget between £200 and £1500 for a pair covers the spectrum from good to excellent. Brands such as Behringer, Presonus, Alesis, M Audio, Swissonic, JBL, IK Multimedia, KRK, Mackie, Fostex, Yamaha, Kali Audio, Adam, Genelec and Neumann offer products falling in this range and there are many more besides.

DIGITAL AUDIO WORKSTATION SOFTWARE

You're going to need some form of sound editor or digital audio workstation (DAW) software for working with your field recordings and there are many options. While this book isn't the place for in-depth software tutorials (there are many excellent online resources ranging from free content on developer's websites and YouTube tutorials to paid-for content at sites such as Lynda.com and Skillshare.com), a DAW will enable you to edit recordings to length, add fades to the beginning and end of tracks, adjust signal levels, create sound collages, process audio into entirely new sounds and lay-up field recordings and sound effects to video. And of course, you can record material directly into the software using microphones and instruments.

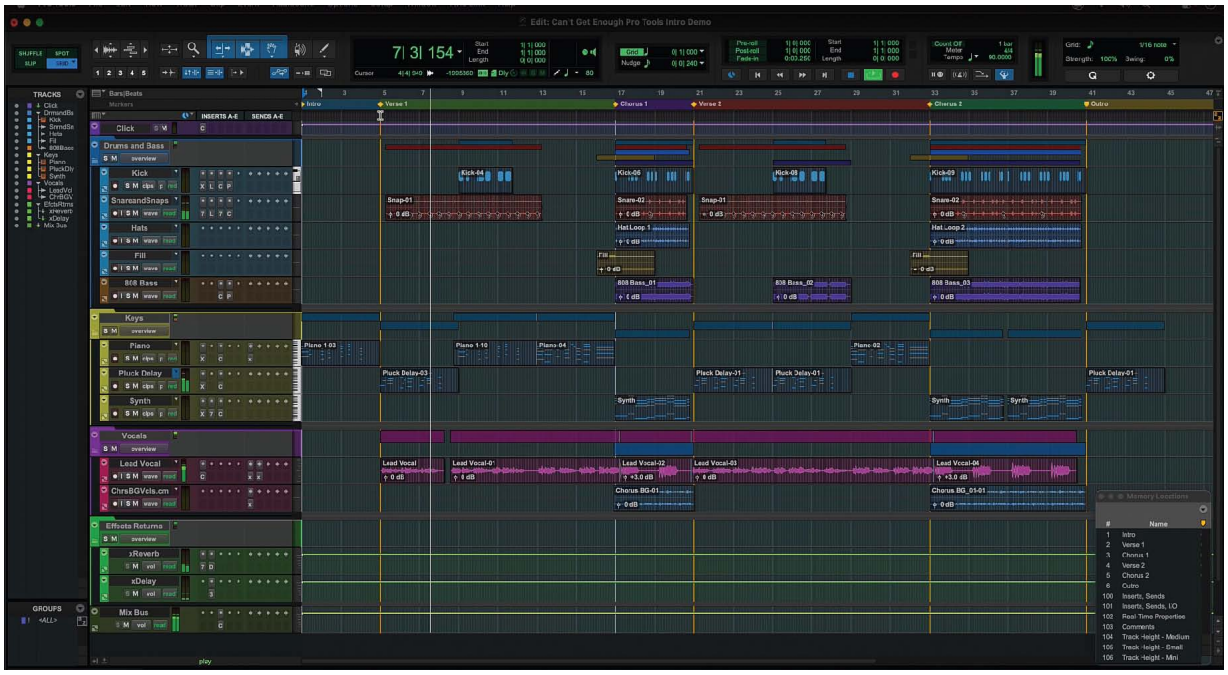


Audacity is a great cross-platform freeware tool used by loads of field recording enthusiasts for basic editing, processing and composition.

One very popular option with field recordists is Audacity. It's free, open source cross-platform software that's been around since 2000 and continues to be updated and developed. It doesn't have all the features of a fully blown DAW but can record multiple tracks if required and is very easy to use for basic cut-and-paste editing, adjusting levels and adding fades. It also ships with an extensive library of plug-ins (more on this subject shortly). Audacity can be used to convert files from one format to another and for the conversion of analogue recording (such as tapes and records) to digital files. With tools for noise reduction, audio repair (to manually fix clicks and other noises) and spectral analysis it's a great starting point and may be all you need. I know many people who use other DAWs who started on Audacity – and still return to it on occasion to process audio simply because it has such extensive and familiar capabilities.

Thirty years ago when DAWs were in development, there were significant differences in terms of features and user interfaces. Most have now adopted a left-to-right scrolling edit window (with the notable exception of Renoise [€68] with its top-down paradigm) and all contemporary software offers similar capabilities, so there are no right or wrong choices. It's often the case that people will have some option already available. For example, anyone with a Mac can download GarageBand (free) and people who already have Adobe Creative Cloud will have Audition as part of the available tools. Field recorders and interfaces often ship with an activation code for the 'lite' (free) versions of well-known software such as Cubase and Ableton Live. Pro Tools Intro is also free.

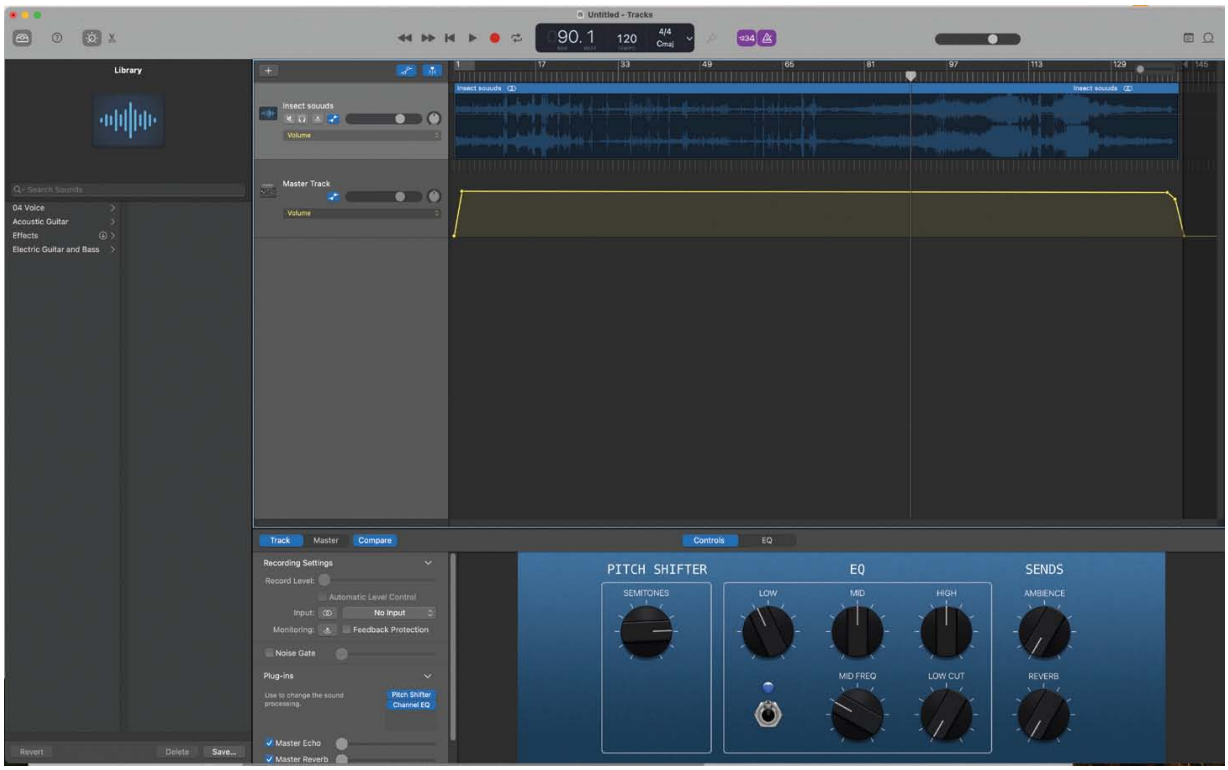
All the lite versions offer high-quality sound but are constrained to the number of audio tracks they can run in parallel, with 8–16 (plus a similar number of software instrument tracks) being the general rule. They also have fewer plug-ins and more limited functionality than the paid-for versions – but many field recordists will find what they offer is more than adequate. However, you will probably need to consider a paid-for version if you are working with multitrack music and sound effects, especially in the lock-to-picture environment. Of the free software, only GarageBand can import video to work with. If you already have Premier Pro, this has some basic sound-editing tools, and I've known many people use these for audio lay-up, editing and basic composition.



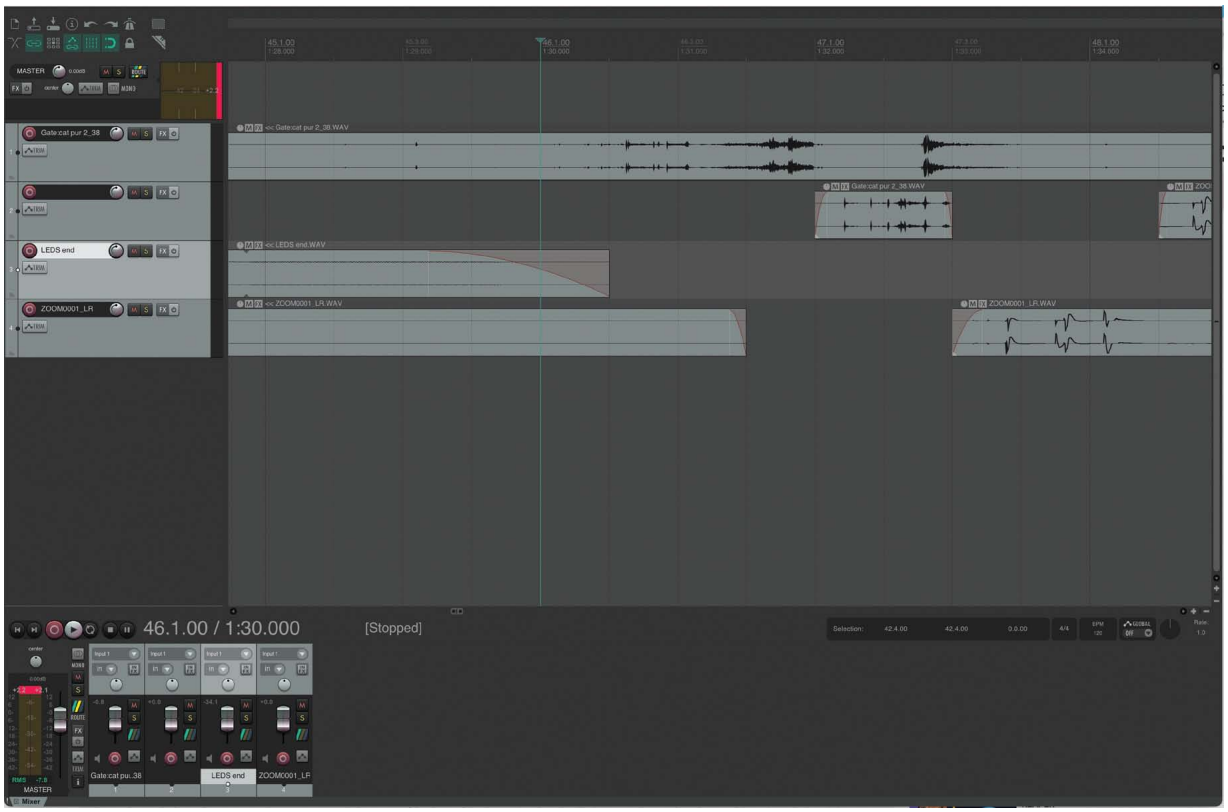
Pro Tools Intro Edit window. Fully featured free introductory version of an industry standard DAW.



Pro Tools Intro Mixer window. Pro Tools was designed to be a two-screen system. It works fine on a laptop but the extra monitor really enhances the workflow.



Garage Band – showing Master Track, Fade and pitch shifter plug-in. Another free software that can meet all the needs of a lot of field recordists.



Reaper. A low-cost permanent licence gets you a powerful DAW with the added benefit of great support for a range of surround mixing setups.

In terms of paid-for DAWs there's a significant range of prices with manufacturers offering different versions of the same software with more or less features. A number of providers have moved away from offering permanent licences in favour of subscription-based models, which reduces the initial outlay. And many manufacturers offer significant student and teacher discounts. Most DAWs have Mac OS and Windows versions – with the notable exception of Logic Pro, which is an Apple product and therefore Mac only. However, the relatively low cost of a permanent licence (around £200), the fact that GarageBand is effectively the lite version of Logic (GarageBand files can be opened in Logic) and the prevalence of Apple computers mean it's extremely popular. Reaper is a powerful, low-cost ($60 licence) option offering both PC and Mac versions, as does Harrison Mixbus (£99.99). As mentioned at the beginning of this section, all the established players in the market offer free 'lite' versions of software, so if you've not already settled on an option there's the

opportunity to check out the competition and see which feels the most intuitive to you. Alternatively, most manufacturers also offer an extended trial period for the full version of their DAWs.

PLUG-INS

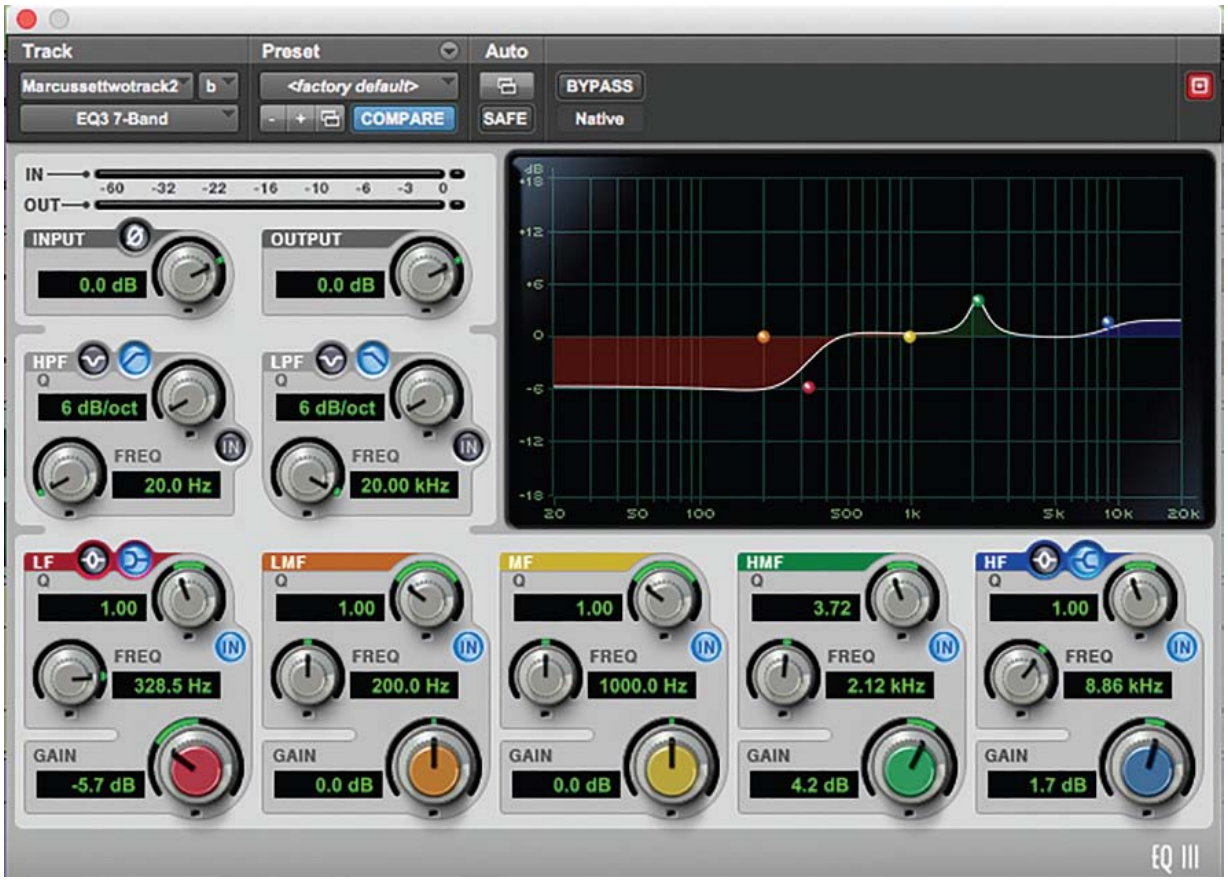
Plug-ins are software components that add functionality to your DAW beyond the basic provision. Fundamental effects such as reverb, pitch shift, distortion and equalisation (tone control) are all applied using plug-ins and there are many designed for more invasive manipulation and processing. All DAWs come with a basic suite of plug-ins and the aftermarket arena is huge. There are many free options for download, budget solutions and suites of products costing hundreds (and thousands) of pounds designed to address every conceivable need. Plug-ins are applied to individual channels, groups of channels or the master channel through which all elements of a track pass. Each will have its own graphic user interface (GUI) and in most cases the GUI will appear as a separate window that can be revealed or hidden as required. In Ableton Live, plug-ins appear in a separate window underneath the main edit window. Plug-ins aren't just static processing tools; all have a range of automation options so you can work in the time domain to create movement and transitions of sonic character. Be aware that there are a number of different formats for plug-ins so once you have chosen a DAW make sure to check requirements. Most plug-ins, but not all, come in VST, AU and AAX formats.

While there are far too many plug-ins to go through specific options, they fall into general categories, some of which will be of more or less use to field recordists and soundscape composers. Here's a generalised overview.

Equalisation

For field recordings, equaliser plug-ins are extremely useful. They are the software equivalent of the tone controls on an amplifier, only much more fully featured. They allow you to cut or boost very specific frequencies. In post-production, the objective is to make the

recording sound as much as possible like it did on the day – to your ears. Microphones do not ‘hear’ exactly as our ears do and some adjustments may be necessary. This is a very good illustration of how important it is to listen to the environment as well as record it. There are many things that can be done with an equaliser plug-in to improve the overall listening experience. Just because there was a significant level of low-frequency rumble from traffic or machinery does not mean we have to live with it; we can dial it out with EQ. Whether we take out all of it or just some is a matter of taste. Many microphones have a bass roll off switch (often called a high-pass filter because it allows only high frequencies to pass through) to take out all frequencies below 80–100Hz, which offers an easy fix. Personally I prefer to record all the information and deal with it in post; sometimes an element of low-frequency content is integral to the experience and you cannot get back what you have not recorded.



Pro Tools EQ 7-band. An equaliser plug-in gives you full control over frequency manipulation. Great for removing unwanted traffic rumble or adjusting bright recordings to sound more natural.

If you are layering recordings you may want to remove certain frequencies from some elements of a track so as not to ‘muddy’ the overall sound-image with too much of a build-up. Or perhaps you need to brighten a bell sound to make it stand out against the background. Generally however, I find the subtractive approach the most rewarding; the less you intervene with the important sounds the more realistic they will appear. If a sound will still not emerge clearly from a mix I will explore its frequency content – boosting different bands – until I find where the problematic frequency is. Then I will reduce or remove it entirely. As you are effectively removing energy you may need to increase the track gain to compensate.

Beyond basic sound management, equalisation can be used in both structural and creative ways. For example, you can make a

voice sound like it’s being recorded over an old telephone or give a general ‘vintage’ tone to a recording by lifting the mid-range and reducing high-frequency content.

Noise Reduction

For many field recordists the idea of being able to remove the troublesome background noise created by budget recorder preamps, faulty microphones, air conditioning, electrical plant and other sources of ‘interference’ is an exciting possibility. However, a degree of caution is required; all noise reduction software/plug-ins operate by removing frequency content and these frequencies will be present in your ‘good’ audio as well as the ‘bad’ noise – the software doesn’t discriminate and too much subtraction can lead to unnatural-sounding results, even if the offending sound source has been eliminated. Aggressive de-noising can also introduce digital artefacts that make recordings sound unnatural. Thankfully, all de-noising software has parameters that can be adjusted. There is always a degree of compromise to consider, so noise minimisation rather than complete removal is often the better route to take. Bertom Audio’s Denoiser Classic is a great starting point for exploring the territory. It’s cross platform and the older version (for Mac OS 10.9–10.12) is free while the latest version comes with a request for just $9.99. There is also the more fully featured Denoiser Pro for $25. It has a deceptively simple interface, but by increasing the threshold of the default preset and then adjusting the individual frequency bands, it’s possible to manage many sources of annoyance.



Bertom Denoiser Classic noise reduction plug-in. A free plug-in that can help you deal with basic noise issues in your recordings.

The majority of de-noising software functions on the basis of training the software on an isolated incidence of the noise, so it's important to try to capture some 'silence' where the noise is the only (or at least major) sound present. This is easy if you're working with a noisy guitar amp in the studio but less so in the field. Having trained the software on this section of a recording, the subsequent profile can be applied to a whole file. Once the noise profile has been generated it can be applied and adjusted to give the best results. If there is more than one noise source in a recording you can also isolate that and make and apply an additional profile. There are creative applications for de-noising as well: if you are working with coil mic and other electromagnetic materials with a lot of mixed frequencies in different but distinct bands, it's possible to separate these if you have moments when only one sound is present to train the software with.

Of course, narrowband noise (such as preamp hiss) isn't the only issue you may need to deal with. Clicks (analogue and digital), crackles, pops and hums are all out there waiting to spoil your listening pleasure. Waves Restoration package ($430 – but often promoted on massive discounts that see it on sale for less than $100) contains five plug-ins capable of dealing with most issues, including two that gather noise profiles for subtracting frequencies. Izotope RX is often referred to as the industry standard for noise reduction and the top-end Advanced (£1149) package contains many excellent tools for audio post professionals that go way beyond the basics, including a de-wind module.



Izotope RX de-noise in progress. You can use RX to remove things like preamp hiss. It can also be used creatively to split up mixed frequency soundscapes into individual elements for mixing and composition.

For those with less demanding requirements (and budget) there's the Izotope Elements package (£114) and the Standard Package (£399). Elements features five plug-ins (de-click, de-clip, de-humm, de-reverb and voice de-noise) plus the new Repair Assistant (common to all packages) that uses machine learning to recognise specific problems and propose a repair chain. This works very efficiently with dialogue but requires some tweaking via the simple interface for other sources. The Repair Assistant has Voice, Music, De-click, Percussion and Sound Effects modes and the latter is suggested for 'ambiences'. However, there is no dedicated field recording mode. Realistically, most of the field recording specific tools, such as Spectral Repair (which can remove wind buffets, thuds, thumps and other annoying noises) and Spectral De-noise, which has broad application for removing system noise generated by preamps or microphones, come with the Standard or Advanced packages. These packages also feature a stand-alone audio editor, which has been a core utility of RX since the beginning. This allows you to work with a spectrogram of your recording, which enables you to see exactly where problems occur and the sonic effect of the processing you apply.

Compression and Limiting

Compressor plug-ins control dynamic range, managing the differences between the quietest and loudest signals in a recording. They are used extensively in music mixing to smooth the dynamics of individual instrument and voice performances and as part of the mastering chain designed to deliver an even-sounding track and maximise the audio output level without clipping. With field recordings, natural dynamics are usually part of the listening experience we will want to preserve and field recordings are best enjoyed at something close to their natural level. However, there are times when we are dealing with recordings that evidence very loud and very quiet sounds where we may want to control the peaks and make the most of the subtle moments.

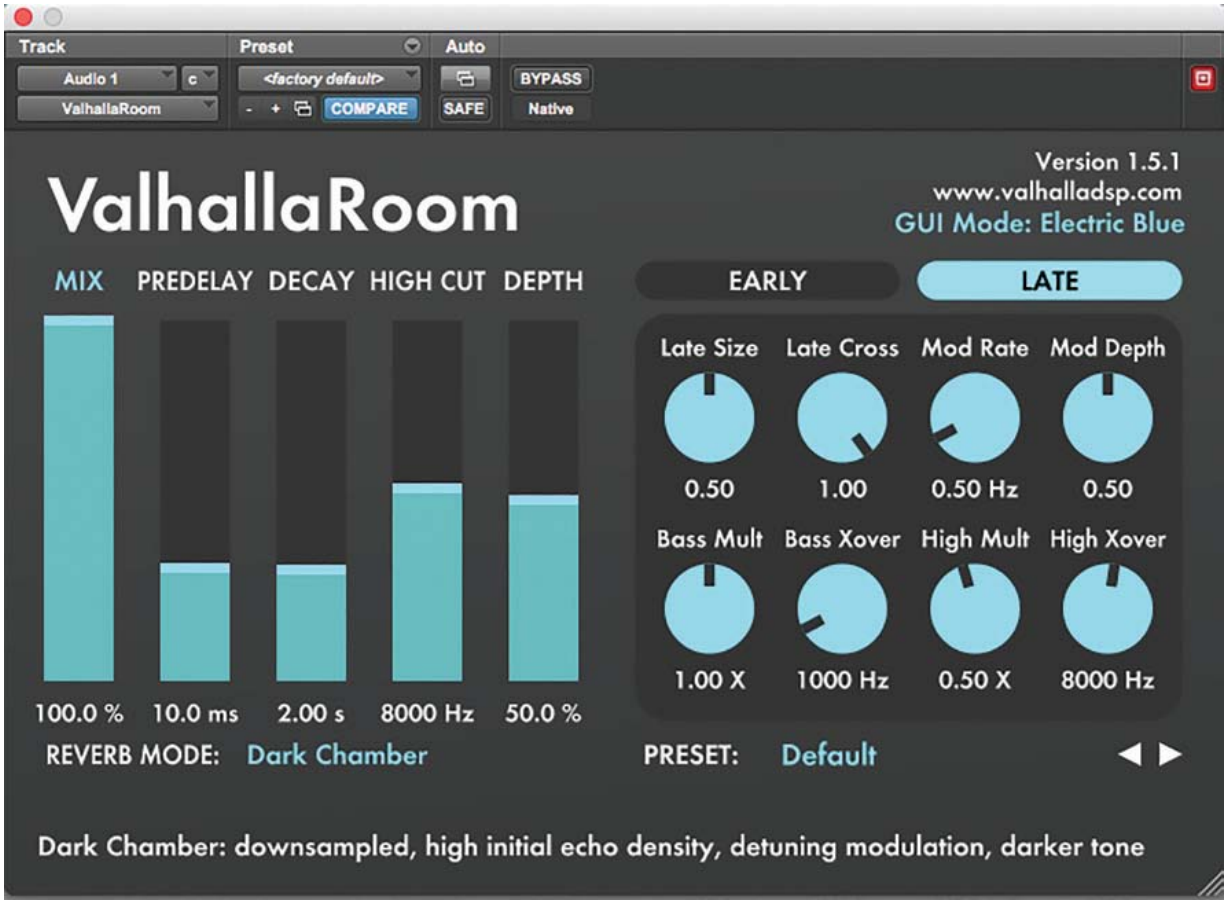


Logic compressor. Like a super-sensitive automatic level control. Used carefully, compressors can bring up the level of quiet background sounds.

You may recall Chapter 1's caution against using your recorder's automatic level control (ALC) because the effect can be quite brutal, leading to unnatural-sounding results. A compressor plug-in can act like a very subtle ALC; if set properly its influence should hardly be noticeable. On the other hand, if a track has a generally healthy signal level but there are occasional very loud peaks, an audio limiter may be the preferable solution – this will impose a threshold, or ceiling, above which no signal may pass. In this way you can keep the track's overall level up and allow the limiter to act only on the very loud peaks. Understanding and using compression and limiting takes time and while presets can be useful, learning to manipulate the various control parameters is the way to achieving the most natural-sounding results.

Reverb

The idea of adding artificial spatial characteristics – reverberation – to a field recording might seem like a perverse idea; generally speaking, part of the beauty of field recordings lies in the way they capture natural ambiences. However, there are times when you may want to add reverb, especially if you are working in audio post with images and you need to place close mic'ed sound into a different environment from that where it was recorded. There are fundamentally two different types of digital reverbs.



Valhalla Room reverb. Artificial spaces can be used to enhance elements of a mix and make them stand out against the background sound.

Algorithmic reverbs (all DAWs have these and there are many third-party options), which use mathematical modelling to emulate

the characteristics of vintage hardware reverbs that used metal springs and plates to create reverberation effects or analogue circuit-based simulations of physical spaces such as small rooms and concert halls. Some are quite realistic but generally speaking, they sound more like effects than actual spaces – which can be useful for sound design.

Then there are convolution reverbs (such as Waves IR1, Altiverb, LiquidSonics Reverberate or Voxengo Pristine Space) that mathematically model real-world spaces and environments. Convolution reverbs come with a wide range of modelled spaces and you can also programme your own. The process requires an impulse response from the actual space you want to model. There are a number of ways to record this: popping a balloon or using a clapperboard, a starting pistol or, more accurately, a sine tone sweep broadcast over a speaker. The recording captures how the space responds to the sound and the impulse response is then imported into the software. Once processed you can apply your reverb to any sound, effectively placing it into the space you have modelled. The results are surprisingly believable.

Pitch-Shift, Varispeed and Time Stretching

In the analogue domain, pitch shifting is achieved by speeding up and slowing down the travel of a recording medium such as magnetic tape. This has both benefits and disadvantages; field recordings can be transformed into entirely different sonic material. The effect can be achieved digitally by using varispeed and all DAWs have comprehensive tools for achieving this. However, if I really want to slow a sound down a lot I still prefer to transfer the sound to analogue tape, manipulate the speed/pitch and then transfer it back to the computer; extreme digital time/pitch adjustment can introduce artefacts whereas tape delivers smooth, even manipulations every time.



Logic Pitch Shifter. Many everyday sounds develop unique creative potential when we start to manipulate pitch – especially when we slow things down.

Pitch shift plug-ins do what the name suggests, raising or lowering the pitch of a sound and they can do this without changing the speed of the recording. While there are obvious applications for computer music making, such as adjusting the sonics of kick drum sound, they are also extremely useful when processing field recordings in the context of sound design. Pitching an entire environment down a fraction can create a sense of foreboding, a certain ‘something’s not quite right’ feeling without being obvious. Increasing the pitch of low metal tones recorded can deliver the sounds for aliens or ghostly swords.

Vibrato plugins can also be considered here as they function by allowing for very slight modulations in pitch. Tremolo on the other hand is achieved using amplitude modulation.

Time stretching is another useful tool. This involves manipulating the speed of a recording without changing its pitch. This is very useful in music where you are using loops and samples that are at a

different tempo to the track you are working on. In soundscape composition, time stretching can be used if you want to adjust event sequences in one sound file to coincide with timings in another.

Distortion

Most of the time field recordists will want to keep distortion out of the picture but sometimes a little grit can be useful when exploring sound for composition or developing sound effects. There are many different distortion plug-ins – and the effects are very different from the clipping we hear in field recordings when the level exceeds 0dBFS. With lots of parameters to adjust such as drive, output, tone and wet/dry mix we can manipulate the amount of break up and the character of our base sound to deliver sonically interesting results. Used in combination with digital pitch shifting and time stretching it can enhance the degradation of sound in ways that emphasise rhythmic, sequenced pulses. It’s very effective when applied to coil or contact mic recordings.



Garage Band Distortion II. Distortion can be useful when you want to add some sonic grit to a mix.

Echo and Delay

These time-based effects have a direct relationship to reverberation; below 0.1 seconds the human brain does not perceive reflected sound as separate from the impulse that created it. Above this threshold we start to identify discrete echoes. What's the difference between echo and delay? While it's by no means an official definition I always think of echo as seeking to mimic real-world sonics with all the characteristics of duration, repeat rate, fade and natural tonal degradation.



Valhalla FreqEcho. Plug-in tools like this enable you to create entire worlds for sound effects.

Delays, on the other hand, explore all the possible processing potentials. Delays can be fine-tuned to have repeats that are louder than the original impulse, offer infinite sustain and be timed to coincide with beats or other repetitive structures. Multiple delays can be layered and delays can be driven into feedback for dub-style and sci-fi effects. All DAWs have delay plug-ins as part of their suite of native tools and there are many free and paid-for options to consider. Valhalla's free offerings include the Freq Echo that combines an analogue echo emulator with a frequency shifter. They also have the Super Massive delay, which also offers reverb effects. The Valhalla Delay ($50) encompasses many types of analogue delay emulation, contemporary digital sounding solutions, reverbs and pitch shifting algorithms. Many developers offer lite versions of their paid-for plug-ins for free. Cross platform options include Imaginando's DLYM, HY-Delay 4 Free, Musical Entropy's Spaceship Delay and Glitchmachines' Hysteresis.

Adding realistic echoes to close mic'ed recordings places them in a simulated relationship to space and using emulations of analogue

devices renders them more like musical materials. The use of complex delay lines takes this investigation of potential even further. Slowly adding delay to recorded ambiences can generate a dream-like quality as the quotidian morphs into something quite other.

Chorusing, Flanging and Phasing

When the delay between a sound source and the reflection of the sound is below 20ms, we experience effects ranging from subtle tonal colouration to more extreme sweeps of frequency. These are the result of something called comb filtering. On location these are characteristics we desperately try to avoid or at least minimise. In the studio however they can be manipulated to creative advantage. All these plug-ins function by creating a copy of a signal that is delayed and treated before being mixed with the untreated signal. The modulation of the delay time creates a moving landscape of subtle pitch changes, which are often referred to as ‘fattening’ a sound. With chorus effect, the delay time is usually between 15–35ms. For flanging effects, which are characterised by a pronounced sweep through a frequency range, the delay time will be much shorter: 1–5ms. Phasor plug-ins also duplicate the original signal but instead of delaying it by a meaningful amount, phasors invert it by moving the position of the copy along the x-axis – the direction of travel – by a tiny amount so that the waveform of the original signal and the copy are no longer in sync. Modulating the phase then allows the second signal to move relative to the original to create interesting, shifting comb-filtering effects as a result of frequency cancellation and reinforcement. All DAWs have on-board chorus, flanging and phasing plug-ins and there are both free and paid-for options to add to your plug-in suite.



Blue Cat's Audio Flanger.

Spectral Plug-ins

Here's where the real sonic mayhem starts! These can broadly be labelled as tools for sound design but many are specifically designed for working with real world sounds. Spectral processing of sound involves its conversion into discrete frequency units. This is achieved mathematically using a process called a Fast Fourier Transformation (FFT). These frequency units can then be displayed using a spectrum analyser. Because the audio has essentially been broken into independent sine tones, very specific interventions can be made. Once processed, audio is returned to the real world by a reverse of the FFT process. This is now being used in all sorts of applications from equalisers to convolution reverbs. A good place to explore the potential of spectral processing is with Michael Norris's Soundmagic Spectral package (AU format only – so suitable for Garage Band, Logic Pro, Ableton, Max/MSP and Digital Performer users). This contains 23 plug-ins capable of mangling sound in every conceivable way. Ina GRM Tools Spectral Transform (€149) is a bundle of 4 plug-ins designed for a range of tasks from the subtle enhancements of existing recordings to complete transformations. Spectral transformation is also used in some stand-alone software, so I'll be revisiting the concept later in this chapter.



Ina GRM Warp from the Spectral Transform bundle. These plug-ins are designed for more invasive processing. Just a single drop of water can be made to sound like an entire sonic world.

Granular Plugins

Granular synthesis is a different but equally powerful processing technique that involves fragmenting audio into ‘grains’: miniscule samples generally of 2–230 millisecond duration. The manipulation of these grains can then be used to create subtle effects or entirely new sounds. Because of its ability to transform field recordings and

real-world sounds, it became popular with electroacoustic/soundscape composers like Barry Truax in the mid-1980s. Today, with some DAWs offering granulator tools (for example Ableton's Max for Live Granulator II or Logic Pro's Alchemy) and plug-ins such as Portal (cross platform – £141) offering real-time manipulation, granular synthesis is no longer the exclusive territory of artists with the computing power supplied by institutional support. The Freeze and Shuffling plug-ins from the Ina GRM Classic Bundle (€149) are also granular devices. As the name Classic suggests, these have been around for a while but the bundle is still highly regarded in sound design circles. Other contemporary tools include Image Line's Fruity Granulator, PolyGas by Stone Voices, Hvoya Audio's Ribs (all free), Native Instrument's Form ($99) and Inertia Sound Systems' Granulizer 2 (€80).



Logic Pro Alchemy. A very powerful tool for manipulating field recording and found sound.



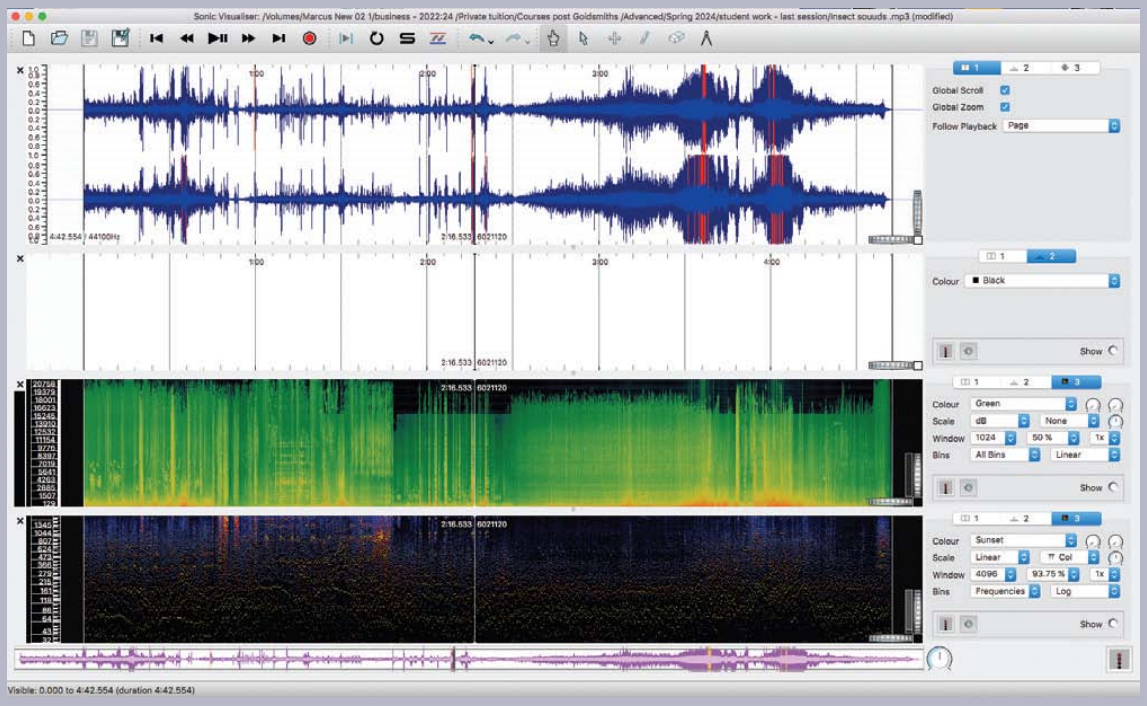
Ableton’s Max for Live Granulator II.

Creative Software

While DAWs are a fantastic working environment there are certain alternative software tools that are extremely useful. Here’s a brief overview of some that I’ve come to rely on.

Sonic Visualiser

Sonic Visualiser is open-source, cross platform software that's designed to provide waveform and spectrogram views for audio files. Images are scalable and text annotations can be added. Audio plays in sync with the image. Sonic Visualiser is really useful if you want to know what's going on in the time/frequency domain.

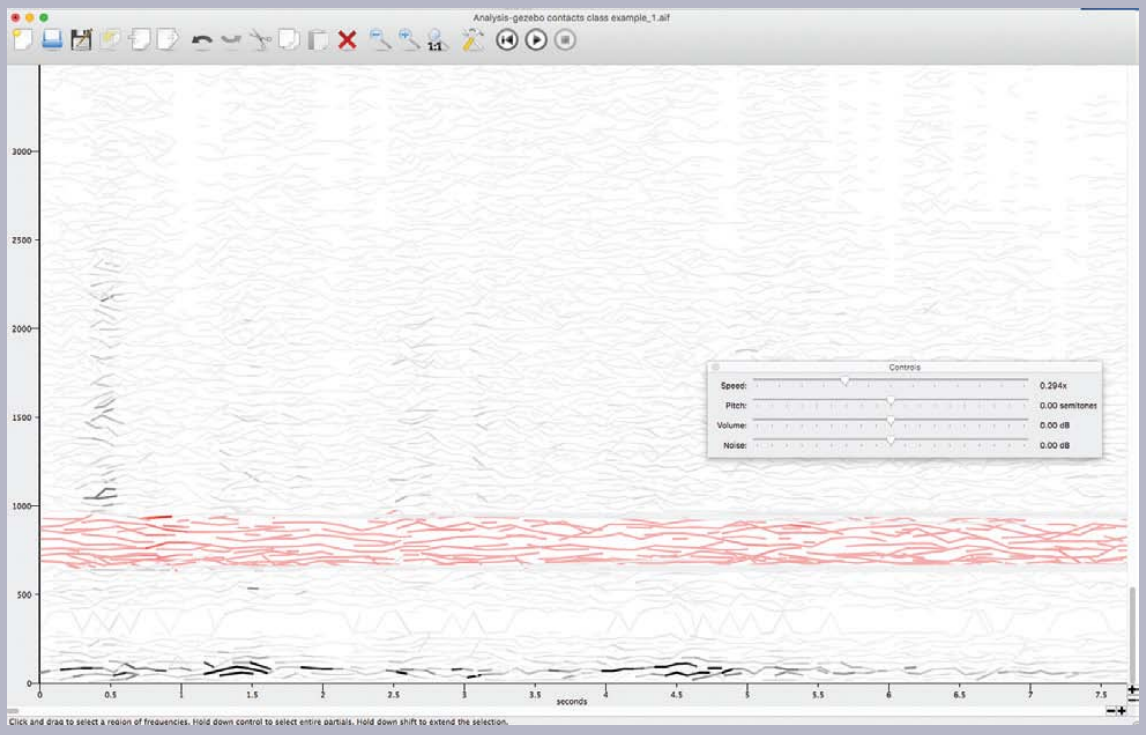


Sonic Visualiser showing waveform, spectrogram and peak frequency spectrogram. This is a great stand-alone software package if you're interested in seeing what is going on in the time/frequency domain.

SPEAR

SPEAR is a free tool for audio analysis, editing and resynthesis. It models sounds using sinusoidal tracks, capturing frequency and amplitude accurately. With field recordings it can produce intriguing, abstract results. Users can alter pitch, speed, volume

and noise, or manipulate individual frequencies. Transformations can be saved incrementally, enabling complex metamorphoses in a DAW.



SPEAR showing application of frequency shift, time stretching and speed reduction. Free cross-platform software with major potential for modifying environmental sound.

Metasynth

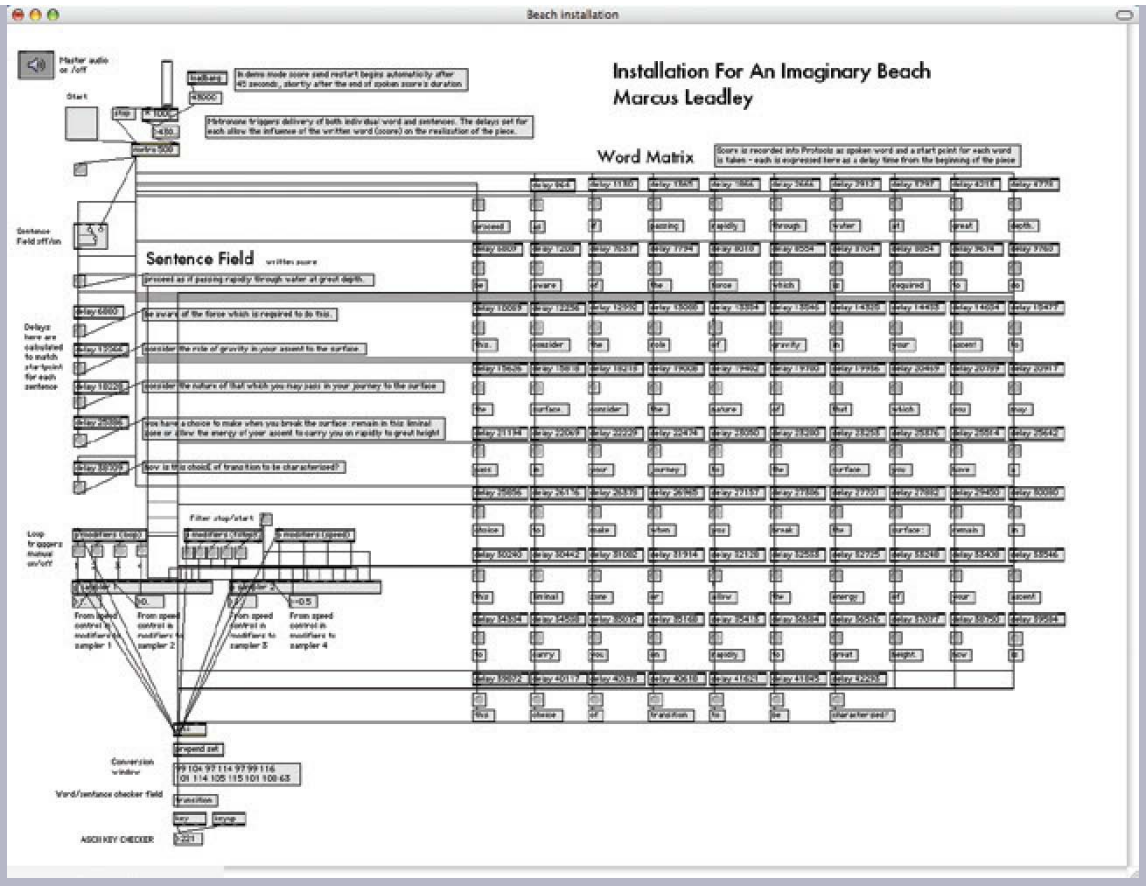
Metasynth is a powerful sound design software used in major films and by creative artists. It enables sound synthesis, image-to-sound conversion and sequencing. The Spectrum Synth Room uses FFT analysis to manipulate partials: ideal for transforming real-world sounds and field recordings.



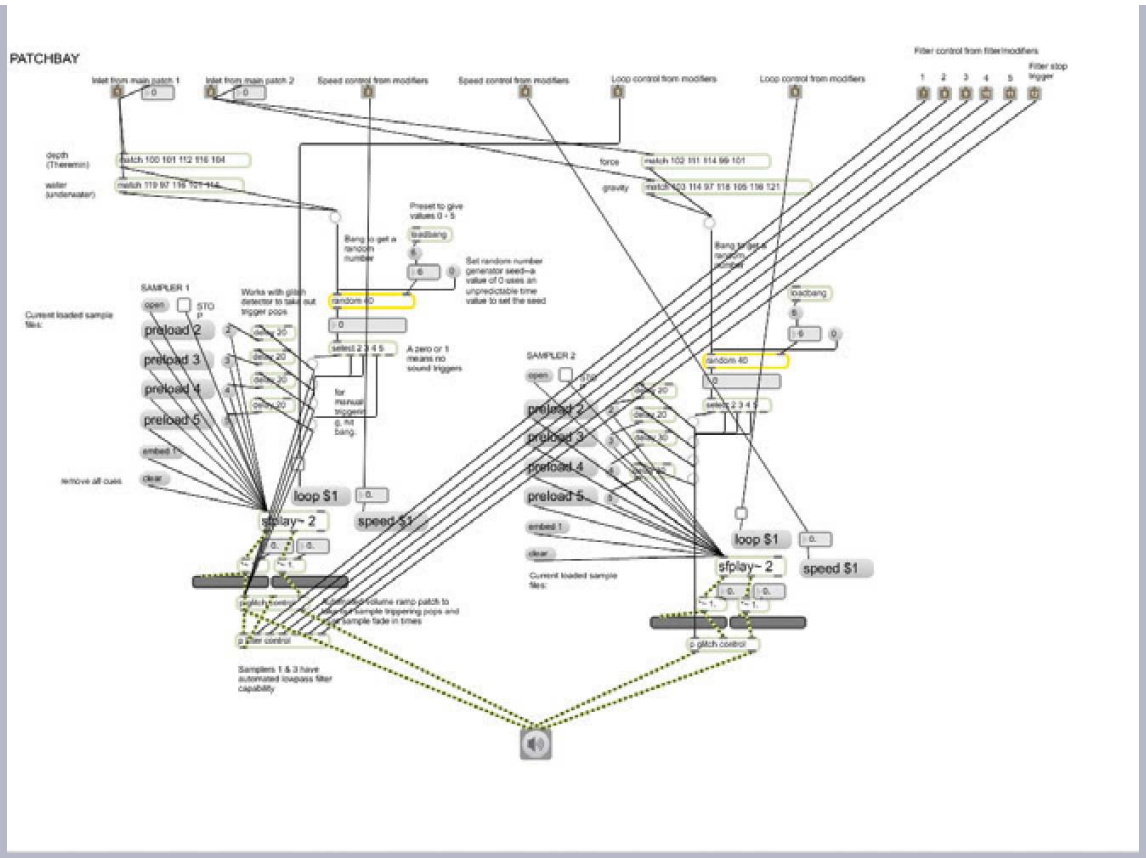
MetaSynth's Spectrum Synth – this can analyse field recordings and allow you to manipulate, filter and rearrange the original's harmonics to create entirely new sonic materials.

Max/MSP

Max uses a visual programming language to build synthesisers, plug-ins or sound manipulation systems. It supports real-time control and includes Jitter (a library of objects specifically designed for working with A/V data). I use Max with field recordings to develop complex alternative soundscapes. Its strength lies in letting users design tools tailored to their creative needs, enabling endless variation and experimentation.



Max/MSP patch designed to recombine short field recordings to generate a mediated version of the sonic environment of any chosen location.



Max/MSP: multi-voice sound file player from within an installation patch.

Useful Apps

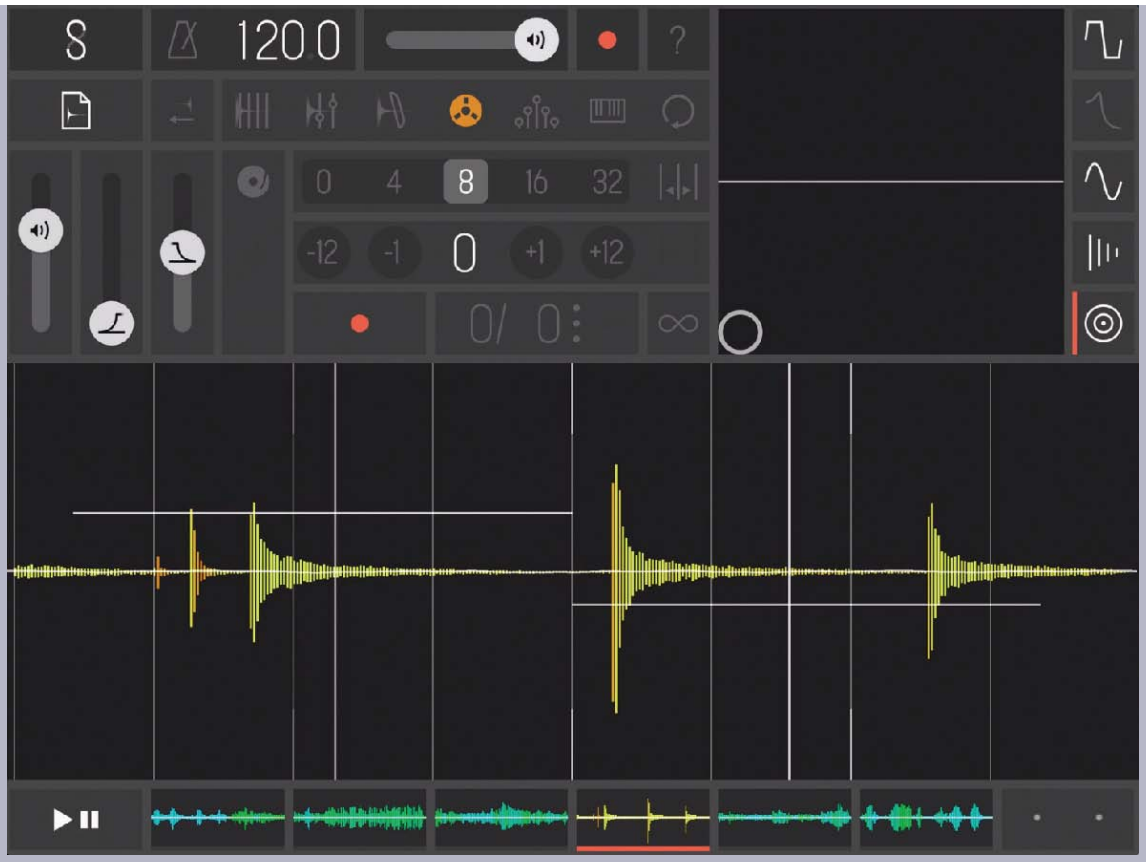
Laptops and desktop machines have physical limitations and the touch screen tablet environment offers a wholly different paradigm for real-time sound manipulation. Here are a few I find intuitive and useful.

GarageBand for iOS
While this is superficially focused on making it super easy to develop musical ideas, it's fully capable of importing field recordings, which can then be trimmed and arranged.

Ableton Note
This lets you sample your environment and is a great way to collect found sounds with your iPhone. You can create sets of sounds and loop, layer, add effects and musical elements. You can leave it at this or transfer the files directly to Ableton Live to continue development.

A DAW for your tablet
If you want a basic DAW to use on a tablet, both FL Studio Mobile (Android, Chrome OS, iOS, macOS, and Windows) and Steinberg's Cubasis (iOS, Android and Chrome OS) can use a microphone or interface to record audio directly or you can work with pre-recorded sound files.

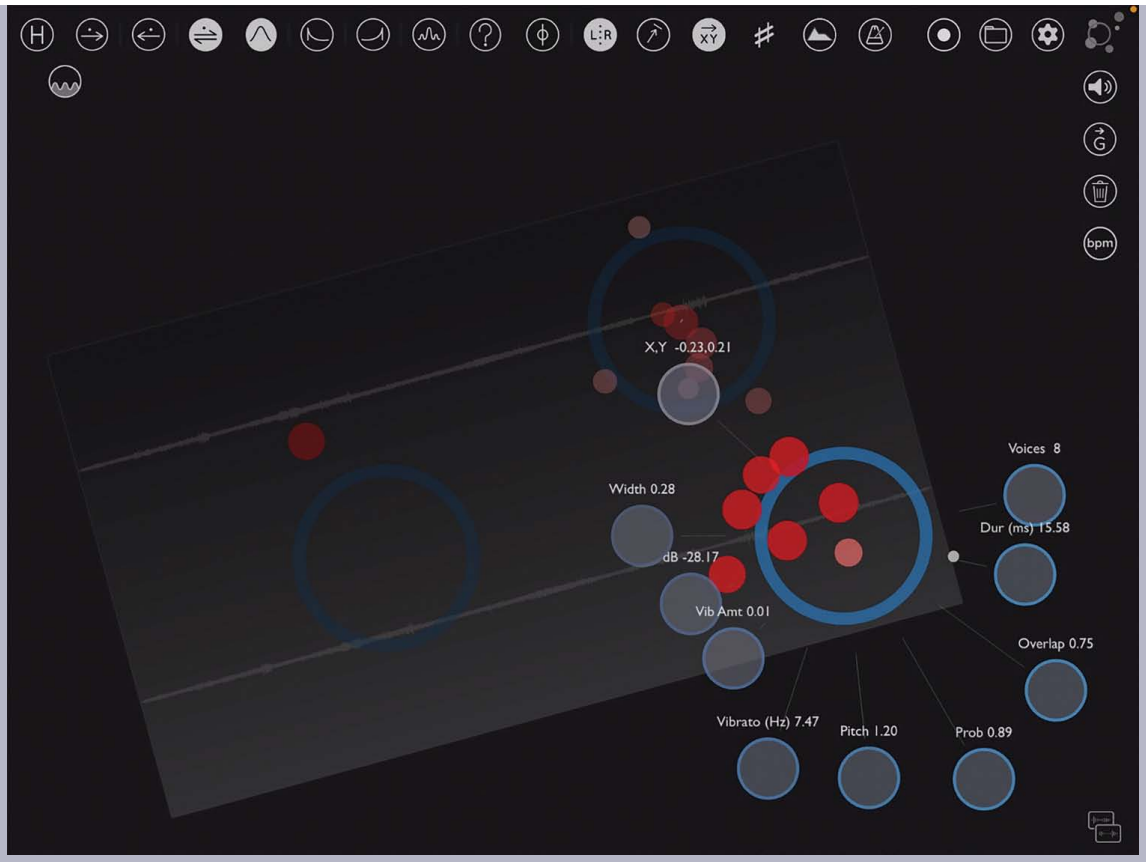
Samplr
Samplr is a versatile app that lets you import or record up to six audio files, manipulate them using eight play modes and five effects, and perform or compose via an integrated sequencer. It's ideal for live performance or studio use with environmental and pre-recorded sounds.



Samplr app for iPad. A powerful, low-cost app that allows you to layer six sound files for processing and playback in real time using various gestural controllers.

Borderlands Granular

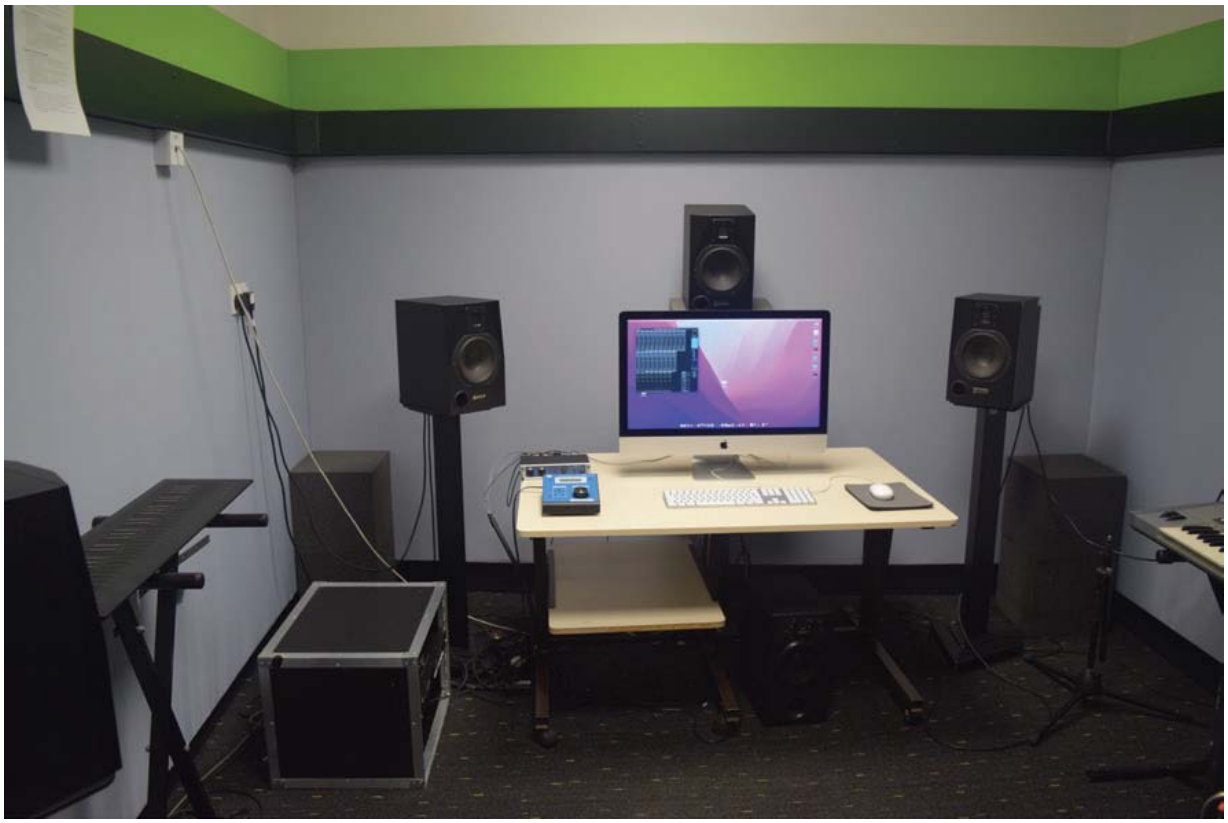
Borderlands Granular is another useful sound manipulator: you can create a ‘landscape’ of sound files and place controllers over the waveform image to control how the sound plays back. Sound files can be moved around to create additional variations.



Borderlands Granular app for iPad. More sonic mayhem for those who like to interact with field recordings as material for live sonic art performance.

HOME STUDIO MULTICHANNEL PRODUCTION

If you’re interested in working with multichannel sound in the home/studio environment you will need to expand your budget, but perhaps not quite as much as you might imagine. The cost of monitors and interfaces has come down significantly in recent years, putting quad, 5.1 or 7.1 sound within reach of many more people. Interfaces with 6 outputs (ideal for quad and required for 5.1) and 8 outputs (suitable for 7.1 sound) can be found at reasonable prices from a wide range of manufacturers. Even where higher output counts are required (9 or 11 are required for Dolby 7.1.2 and 7.1.4 respectively), more products are entering the market place, which inevitably brings down the buy-in price for the technology. Current options include the Motu 16A (£1400), Universal Audio Apollo X16 (£3400) or Focusrite Red 16Line (£3400).



5.1 surround workstation. While a basic 5.1 mixing system will cost more to set up than a stereo system, contemporary sound monitors and interfaces offer great value for money – so you don't have to be a working professional to develop multichannel skills.

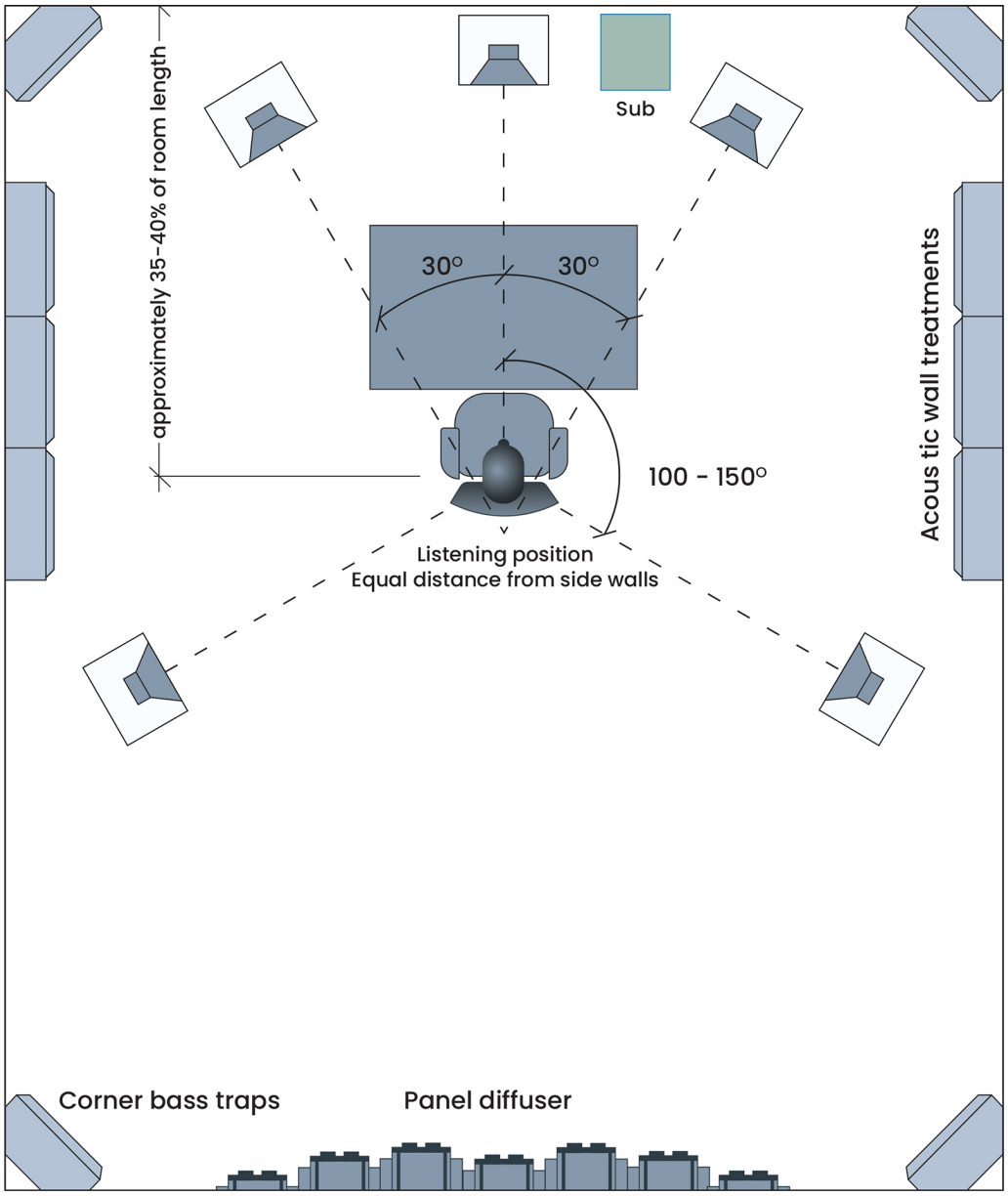
While it's possible to simply send channel outputs direct to speakers, most DAWs now feature a surround output bus and a surround panner. Pro Tools (Ultimate and Studio versions), Nuendo and Cubase, Logic Pro, PreSonus Studio One and DaVinci Resolve Studio all have integrated Dolby Atmos rendering capabilities.

So if hardware costs are falling and software capability is developing apace, how best to explore spatial audio and multichannel sound in the home studio environment? As discussed in Chapter 4, Ambisonics offers a good starting point for field recordists, not least because it can map to any speaker configuration including stereo and binaural stereo for headphones. Hence it's possible to explore and compose without a multi-speaker setup using headphones: part of the Dolby Atmos suggested workflow is to compose and mix using binaural (headphone) monitoring and only hire an Atmos suite for the final production mix.

For those of us who want a home multichannel system, four sound monitors is the minimum. Devices such as the Zoom H3-VR recorder, RØDE NTSF-1 or Sennheiser Ambeo mics record A-format information which is then converted in postproduction using software tools such as the Zoom Ambisonics Player, RØDE's SoundField and Sennheiser's Ambeo A-B Converter to stereo, binaural, quad, 5.1 surround and so on. Once you have the correct files for your setup, you can begin an exploration of spatial audio.

Starting with a 5.1 setup makes a lot of sense, especially for field recording; the workflow from recording to diffusion is an easy path to navigate. For example, loading suitably decoded files into Logic and choosing a 5.1 output bus involves little more time and effort than setting up for a stereo session. The software will instantly distribute the sound across all 6 speakers to give an incredibly realistic rendering of a location. As a surround experience this is quite magical the first time it's encountered and may be all you need to enhance your listening pleasure. Alternatively, placing composed elements and sound effects into the mix and manipulating them is straightforward, achieved by automating the surround panner for each new track to define location and movement. And the format is ideal for working with sound and image compositions.

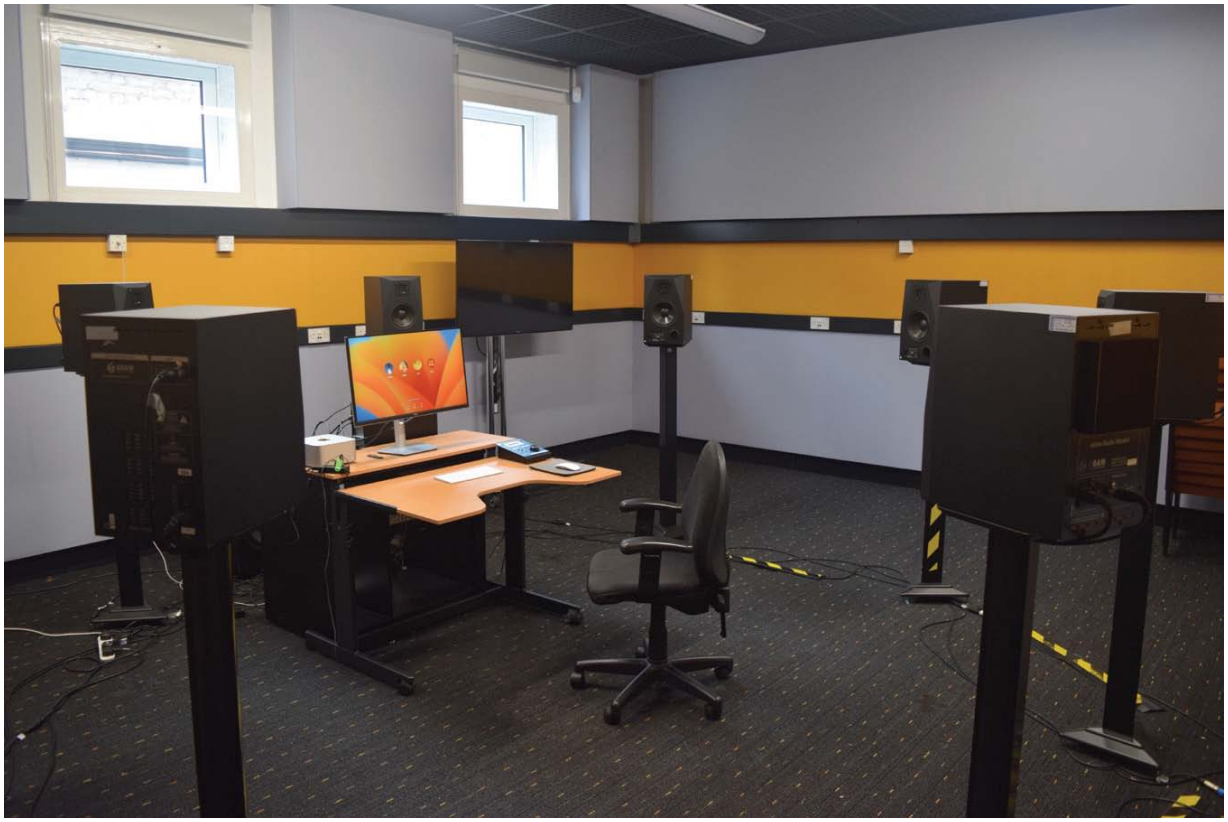
In terms of a workstation setup, it's worth referring to the recommendations laid out in the International Telecommunications Union's ITU-R BS.775-4 for multichannel sound systems. This proposes that the right surround (RS) and left surround (LS) speakers should be placed between 100 and 120 degrees in relation to the centre monitor (*see* diagram). Note this isn't the only option; The National Academy of Recording Arts and Sciences (NARAS) recommend that these monitors be placed from 110 to 150 degrees.



Idealised view of 5.1 audio workstation system setup.

With 5.1 and other surround setups, the sub bass speaker plays an important part in both mixing and creating the immersive audio experience. Where you put it in the room doesn't have a clear-cut solution; low frequency is considered omnidirectional rather than directional and the response may vary significantly as you move around due to the room modes we mentioned earlier. Generally speaking, forward placement on the floor near your centre speaker works well but moving it a metre or more away in either direction

may give a more balanced outcome. Try the sub in a corner as this reduces the potential for null points to form. As with a stereo workstation, corner bass traps and wall treatments may ultimately be needed to manage the room acoustics. If you are still experiencing problems you can ‘tune’ the sub’s location by putting it at the listening position (on your chair) and then you move around the room while playing some familiar audio. When you identify the place where the bass frequencies sound the most balanced (neither under represented or too boomy), move the sub to that location and sit at your chair to listen.



8-channel (octophonic) surround workstation. Moving beyond 5.1 comes with a higher buy-in cost but the environment can be really exiting to explore.

The main disadvantage with 5.1 and 7.1 is that because the formats were developed in relation to cinema, there is a bias towards the specific orientation defined by the centre speaker. Octophonic diffusion, with a ring of equally spaced speakers, is a more open

configuration for environmental sound. It’s possible to do without a sub if your monitors have a good full frequency response, however the low-end driver adds another dimension to surround presentation. Moving to Dolby Atmos production necessitates the addition of either 2 or 4 ceiling speakers or upward-firing speakers to generate the sense of height extension that a suitable 3D panner can create.

Multichannel Composition and Mixing Case Study: James Bulley, Sanctuary of the Unseen Forest

Since roughly 2021 I've moved entirely to doing object-based mixes; it's incredible for spatial work and it changes the way you think about sound. I use L Acoustics L-ISA controller currently, but have previously worked with tools like SPAT, DearVR, BlueRipple etc. L-ISA Controller was built originally for live sound mixing but they've been supporting me to use it for composition and sound design. Essentially you're mixing in an idealised globe – agnostic to what speakers are actually being used. You might be monitoring on headphones (binaural) or 400 speakers in a cube – where they are doesn't matter on the front end. The object-based mix has up to 96 sound objects in it. You just import your speaker system or the desired end result, for example 5.1, 7.1, 24 channels, 8 in a circle, etc.



James Bulley has moved entirely to doing object-based audio mixes in recent years. (Photo: Havva Bulley)

In the simplest scenario I would set the appropriate speaker array in L-ISA Controller and then import my 16 stems as sound objects. These can then be arranged in a circle corresponding to the speaker positions and original microphone distances and I can send the objects to channels 1-to-1. From there I can start to manipulate proximity, height and distance etc. So your 16-channel mix effectively becomes one multichannel sound object. This is very powerful because, if you decide you want a separate recording of say, an owl, to move through this

sound environment you can apply different oscillation to how it moves to create specific results – essentially synthesising spatialisation – with results you simply couldn't achieve with older style surround mixing techniques. You can apply Ambisonic reverb, which is very useful if you have recordings made in a really dry studio. So I will do most of my work like this – even for works that are designed for just stereo speakers. If I'm working on a very bespoke speaker system – for example, 8 speakers over to the left, one central speaker inside a statue and 4 speakers on the floor on the right I can create a CAD drawing of the arrangement, convert it, import it and mix for this setup to scale.

MASTERING FIELD RECORDINGS

Mastering is a term that comes from the days of vinyl records. Vinyl mastering is the complete process of taking the studio or field recording and adjusting the level, dynamics and equalisation so it will perform well with the characteristics of the medium. When people refer to mastering today, they are generally talking about the digital postproduction process of getting files ready for ‘publication’ on a platform such as SoundCloud or BandCamp. Culturally, recorded music is used for creating social ambience and personal immersion in a way that, for the most part, deliberately masks environmental sound. So if your track sounds quieter or less polished than the competition it will lack impact and sound unprofessional. So, for music, mastering can be quite an invasive process involving the application of equalisation, limiting, compression and gain maximisation.



Digital mastering using external hardware. You can do everything you need to inside the computer but hardware can give work a professional sheen.

Unlike music, field recordings are best appreciated at something close to their *in situ* listening level and with all their natural dynamics preserved; this is an integral part of the documentation they represent. In his workshops Chris Watson points out that once we separate sound from source and listen without all the additional sensory content (especially visual), there is a tendency to listen too loud. Jez riley French agrees:

> The best advice is to turn down the listening level; we're so used to listening at too loud levels. Very often the hiss you're hearing is to do with that. People come into my sessions who are working with early Zoom recorders that are really hissy and they say, "I've got this recording I really like but it sounds awful" – and all I do is knock the level back a couple of decibels and most of the problem is gone.

So, while we don't want the listener to have to turn up the volume so much that the noise floor of the recorder becomes audible, field-recording files don't need to be maximised in the same way as music. And there may be other reasons to keep the level relatively low. As well as making binaural recordings, Dallas Simpson was a mastering engineer before he retired. He emphasises the personal nature of the overall recording and presentation process when it comes to level management:

> Virtually all my soundscape recordings use in-ear binaural microphones, so the recordings I make are intimately connected with my perception. In the case of in-ear binaural recording the sound captured onto the recorder is highly complex and has a unique character, which is highly influenced, literally, by the microphone spacing in each ear and the physical conformation of my own head-ear-body shape. This means that the resultant recording has a strong, unique and personal sonic signature [a unique recording equalisation] as well as a strong microphone character imposed by the effect of the ear canal on the microphone capsule, which has to be considered when presenting the final mastered recording for general listening. In some recordings I set my presentation level quite low to reflect the perceptual experience of the location, to encourage the listener to "reach out" with their listening perception rather than have ones' ears assaulted – as in the case of loud compressed music.

riley French expresses a similar perspective:

> In terms of my exhibition work or releases they tend to be unmastered. It's tricky for me and to some extent I've cut my own nose off to spite my face; I don't think reviewers listen for very long to my music because sometimes it sounds completely silent if you listen on laptop speakers. I'm working with frequencies that often don't come out. Sometimes, if

> something's going for a radio show or an external website that's not mine I will master it. I do it myself, and it's all manual adjustment. Gain is the one thing I will adjust in postproduction as I think it's better to under record in the field rather than over record; dynamic range is so massive in the field it's always better to record a bit low. I'll use some basic plug-ins and listen through different speakers. I always make a new copy of the project and master that as, even though in my DAW you can reverse changes easily, I always think it's a good idea to have an unmastered "master" version.

Chris Watson has a pragmatic approach to mastering that always takes into account the destination of the end result:

> If I am asked to supply tracks for film, TV or a radio programme I usually employ a limiter on the main output. I never attempt to master my own work for commercial releases and rely upon the highly expert and experienced services of selected mastering engineers.

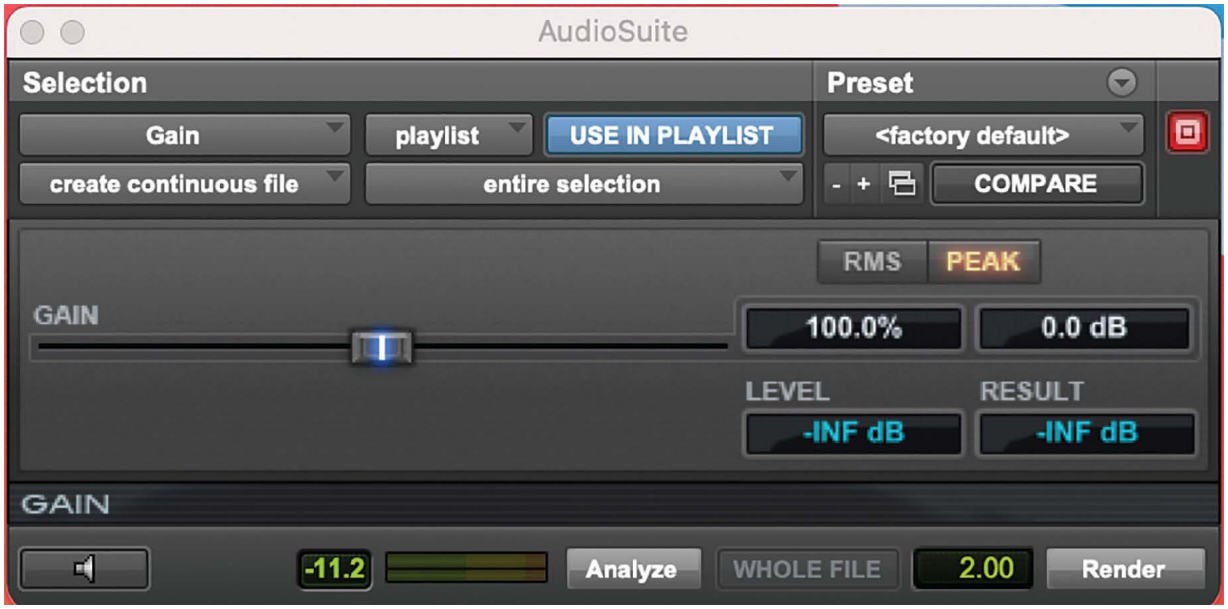
Normalisation or Gain Adjustment?

So what do you do if you want to lift the level of quiet audio files? One solution is to increase the track gain manually. All audio editing software and DAWs can do this. How much adjustment you make is a personal decision. However, there is one basic rule: the adjustment must take into consideration any peaks and ensure these don't exceed 0dBFS – otherwise there will be audible clipping. Hence if there are some loud sounds in an otherwise quiet recording there's little you can do unless you're willing to be more invasive, editing out the offending peaks, using volume automation or limiter plug-ins to manage the louder sounds. Of these approaches, the former will introduce discontinuity and the latter will affect dynamic range. My opinion? In most circumstances embrace the dynamic range and leave things as they are. Alternatively, how important an aspect of

the soundscape is the problem sound? If it's repetitive then it's part of the overall quotidian sonic environment. However, if a helicopter happens to randomly fly low overhead at a certain point in a two-hour recording, how long does your recording actually need to be to capture the essence of the environment to your satisfaction? Ten minutes? Twenty minutes? One hour? If you have what you need, simply end the track and fade before the chopper comes into earshot. Of course, if you're trying to evidence noise pollution your decisions will be quite different.

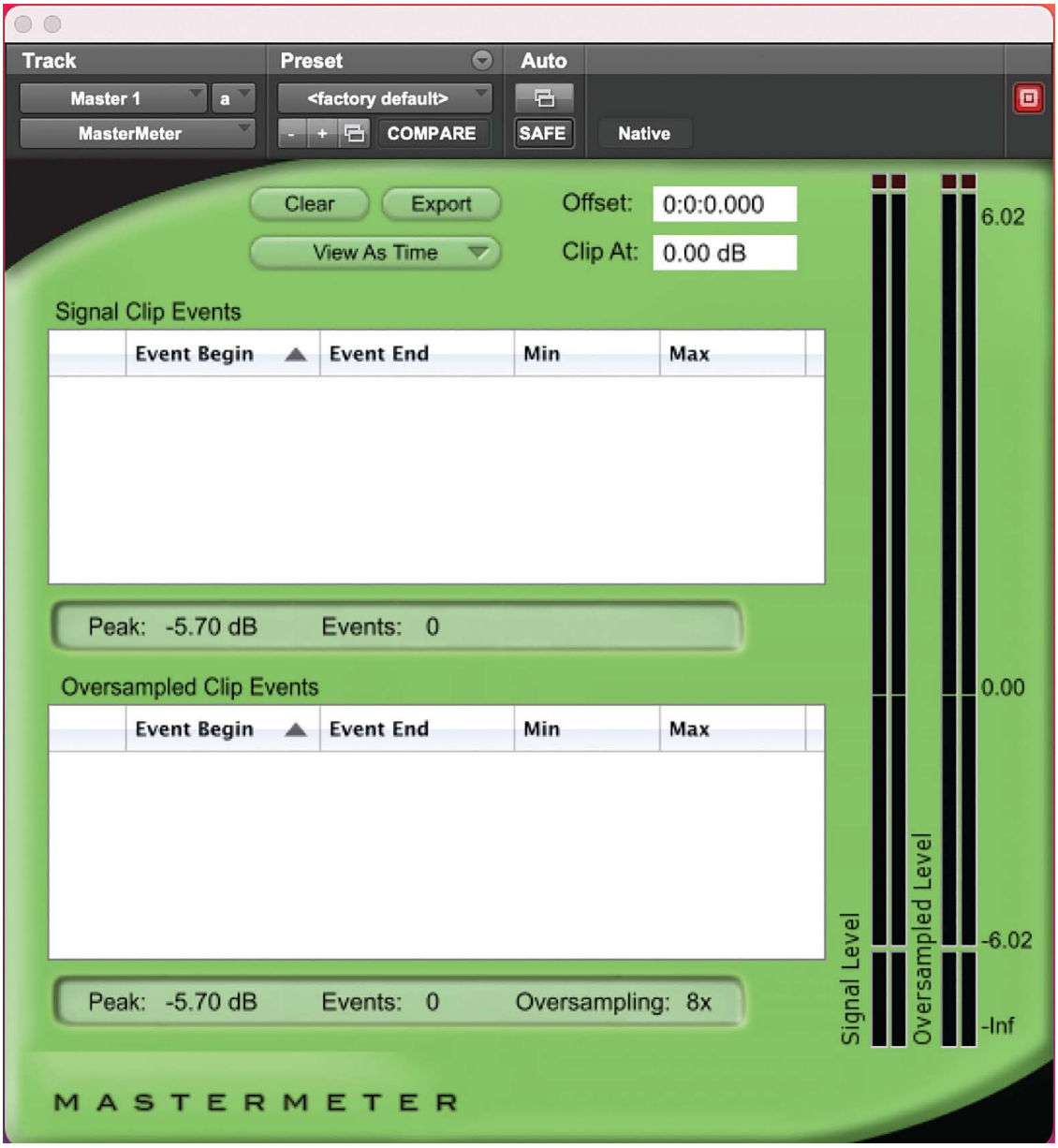


Pro Tools Normalise plug-in.



Pro Tools Gain plug-in.

An alternative to manually adjusting gain is to normalise the file. Normalising is an automatic process that identifies the highest peak level (there is also a version that used RMS [root mean square] averaging which can be appropriate for highly compressed musical materials, but which doesn't play well with the wide dynamic range of field recordings) and then brings all the audio content up so that the loudest peak corresponds to exactly -0.1dBFS. Technically it's more accurate to think of it as a subtractive process as we are working with minus values. Whatever the case, it sounds like a good idea – but in reality, few subjects divide audio producers as much as normalisation.



Pro Tools Master Meter. For most situations your DAW's metering is adequate to ensure you won't be getting any distortion – especially if you stick to a safety conscious -3dBFS.

The arguments against normalisation focuses on a number of issues. Firstly, that it's a destructive process that negatively affects sound quality. The former is certainly true, but the latter is something of a legacy argument: potentially the case in the early days of low-resolution digital audio, less so now – but these ideas stick. Realistically, if you're working with a 24-bit sampling rate or above, and especially with high-resolution 32-bit audio and a contemporary

computer, normalisation isn't going to impact sound quality. And normalising to 0dBFS isn't the only option; setting the 'target' value to a safety conscious -3dBFS makes a lot of sense – especially with what we've been saying about field recordings and sound level. In all honesty, I don't normalise recordings myself; if the gain needs adjusting I do it manually, just in case; old habits die hard.

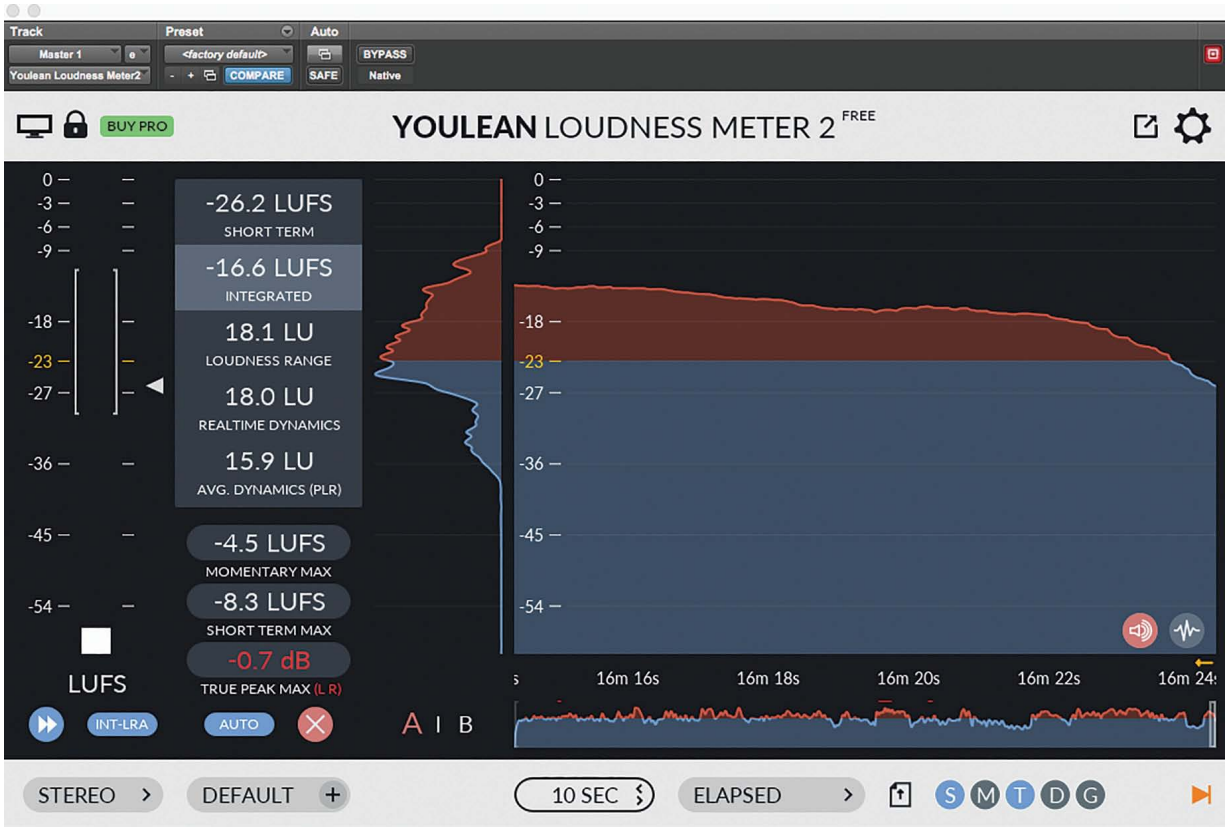
Another issue to consider is the potential for audible inter-sample output distortion when digital audio that's been maximised is converted back to an analogue signal for listening; the signal level between samples may exceed the 0dBFS threshold and result in clipping. This is most likely to be noticeable when lossless audio close to the 0dBFS threshold is converted into a lossy format like MP3 or AAC. So this is an issue to be aware of if you're preparing audio for upload to a streaming service. This is a rapidly evolving service arena and the best advice here is to check with the FAQ section of the site in advance to find out how audio will be processed.

Metering

In order to really explore the issues relating to signal level, we need to be aware that there's a lot more to metering than meets the eye. Your DAW will offer a range of metering options. Alternatively, you can download a contemporary metering plug-in such as the Youlean Loudness Meter (free – pro version £49) or Waves WLM Plus Loudness Meter (£23). And remember that your meter plug-in needs to be the very last thing on the output (master) bus.

Standard digital peak meters only measure the level of individual samples. So, if the sample doesn't exceed 0dBFS it won't highlight an issue. True peak metering is an alternative that takes into account the translation from the digital to the analogue domain and will identify where inter-sample distortion will occur. Peak values are ineffective when it comes to measuring perceived loudness, but useful for material with transient peaks such as percussion of loud natural sound. RMS (root mean square) monitoring on the other

hand delivers a short-term average value that is representative of the overall energy.



YouLean Loudness Meter 2. True peak metering takes into account the translation from the digital to the analogue domain and will identify where inter-sample distortion will occur.

The contemporary standard for metering is LUFS (loudness units full scale). Loudness units are used to quantify the perceived loudness of a track and reference an analysis of an average level over time. Because LUFS references to full scale we're again working with negative numbers so the less negative the number, the louder the audio is. LUFS are standardised (EBU R128 – Loudness Normalisation and Permitted Maximum Level of Audio Signals) and have been adopted across a range of industries so broadcasters and streaming services publish LUFS values for content – either what they expect for content or what they will normalise material to. Spotify for example will adjust tracks to -14dB LUFS. If we look at

the GUI of the Youlean Loudness meter we can see that it provides a wealth of information. Better still, the Pro version incorporates streaming presets for all the major players that can be summoned up as needed.

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7
A HISTORY OF FIELD RECORDING

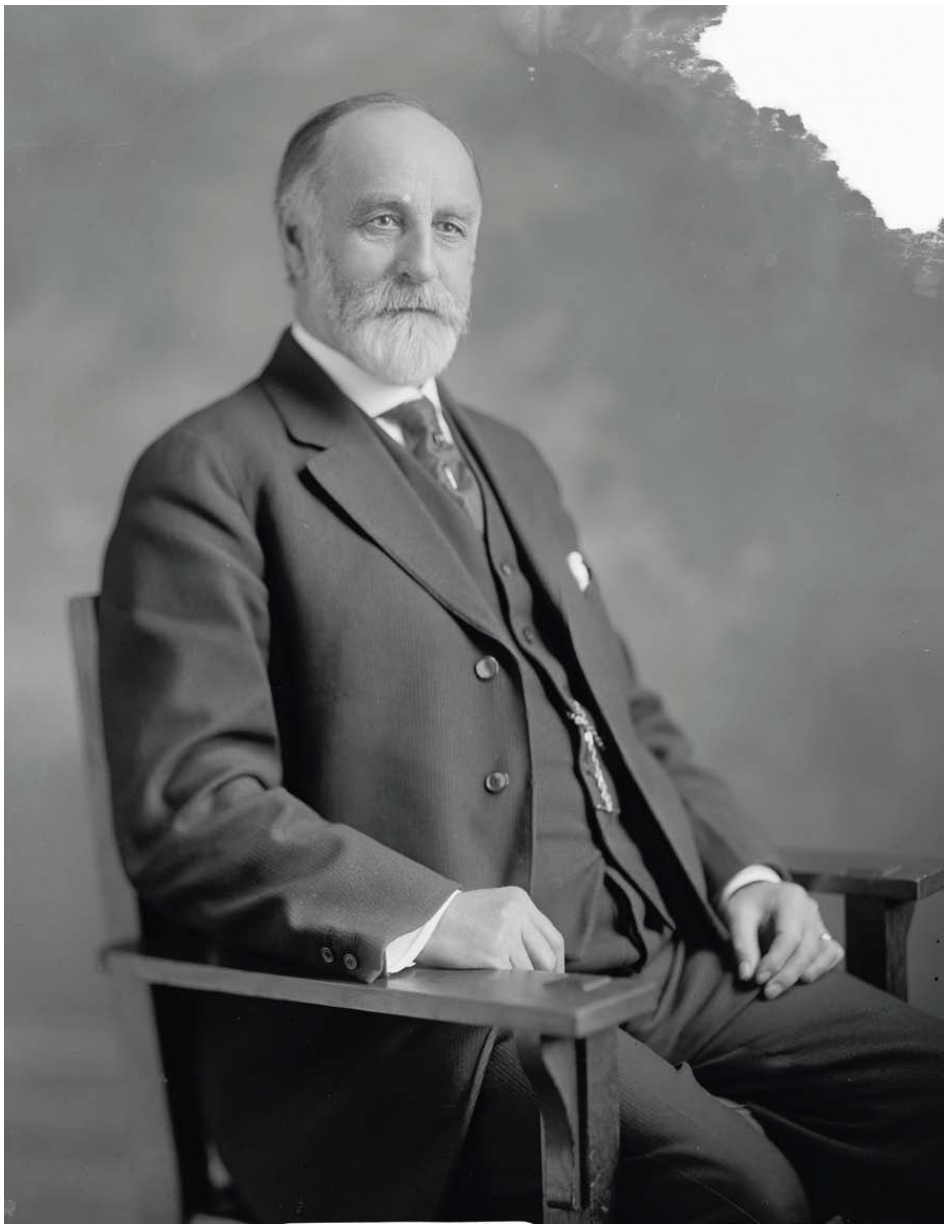
It’s hard to imagine a world in which there is no mechanism for recording sound, and yet this was the case until Thomas Edison invented the phonograph in 1877. While various early efforts were made to capitalise on the patent it was not until the late 1890s that viable machines entered the market.



Ludwig Koch setting up to record in the field during the 1950s. (Photo: Erica Koch, with permission from Anthea Kennedy)

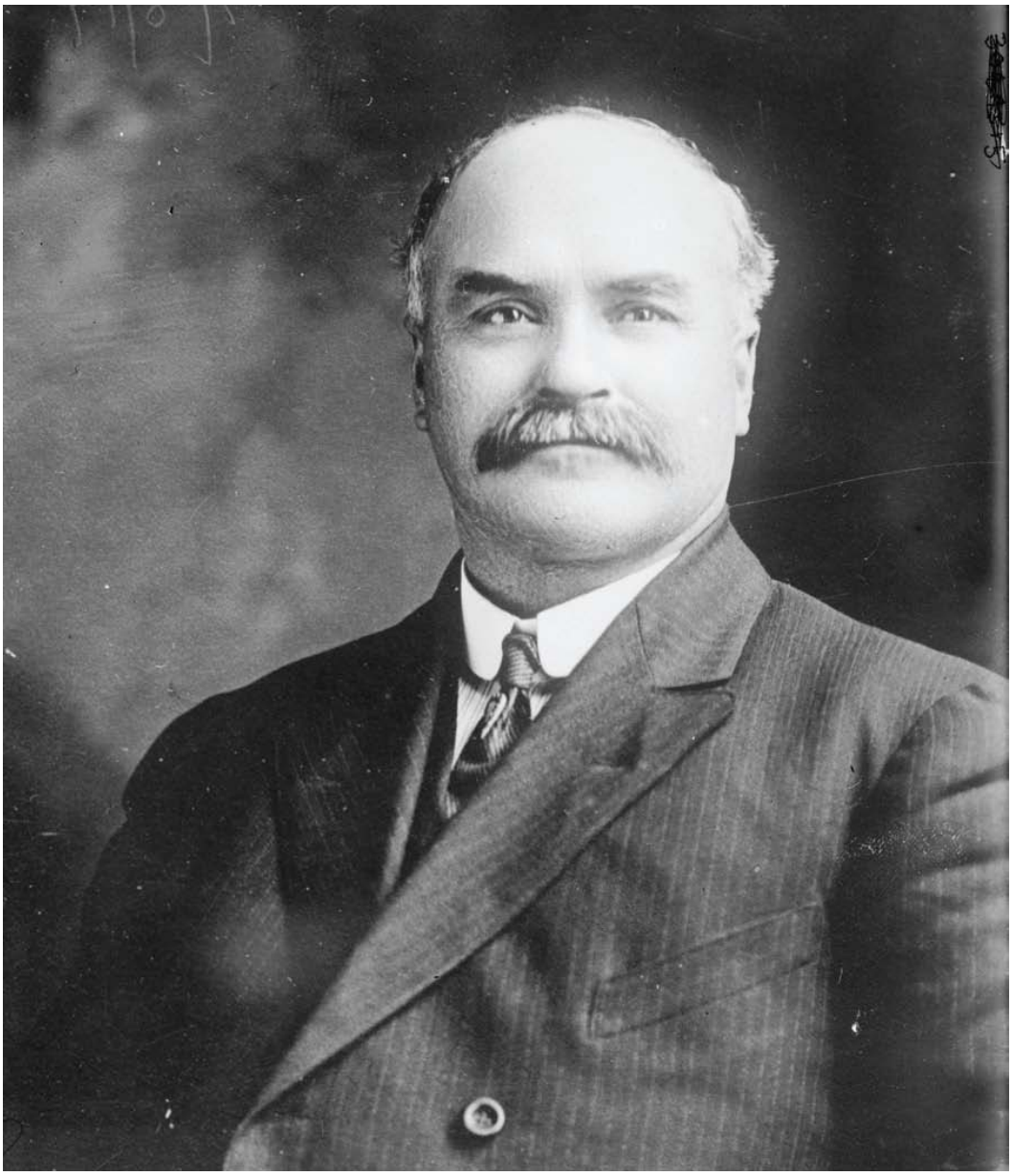


Thomas Edison with prototype phonograph circa 1877. (Unknown author – Library of Congress, Public Domain)



Dr Jesse Walter Fewkes – one of the first anthropologists to experiment with phonographic documentation. (Harris & Ewing, photographer, public domain)

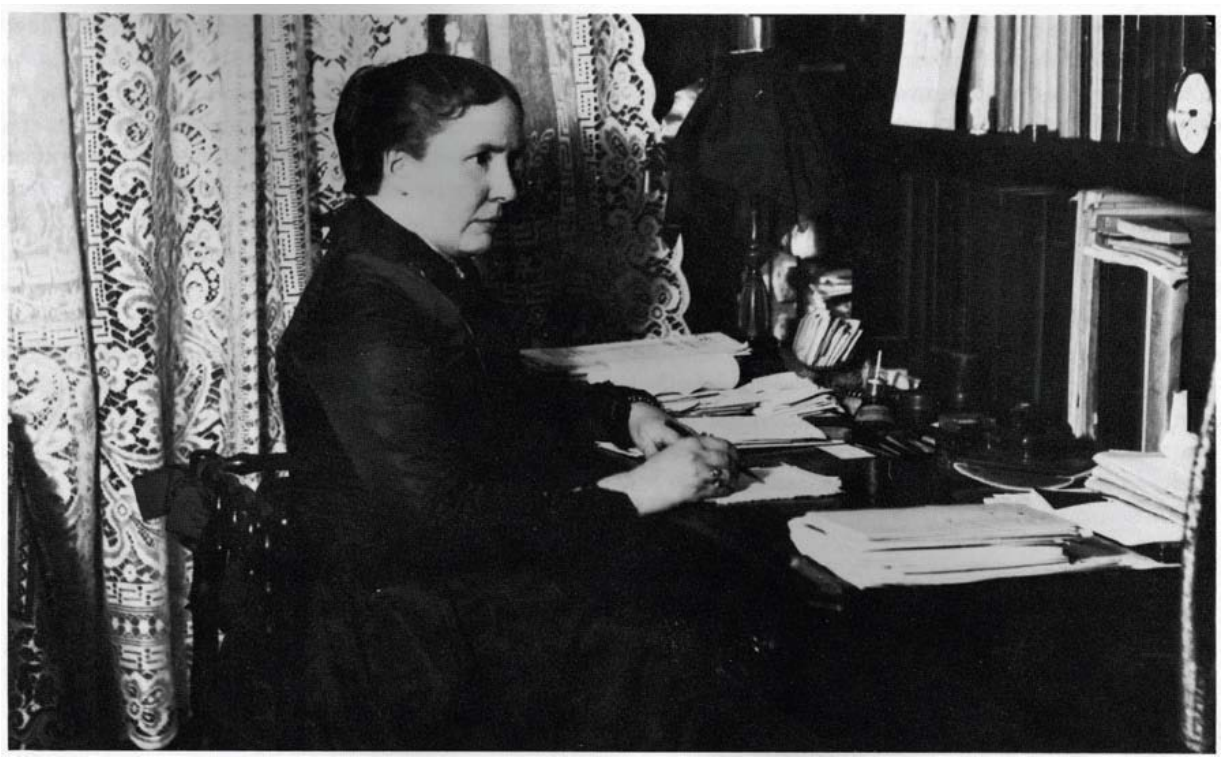
In March 1890, the anthropologist Dr Jesse Walter Fewkes experimented with phonographic recording for ethnographic documentation. He recorded 36 cylinders of language, songs and legends of the Passamaquoddy in the communities of Pleasant Point and Indian Township in Maine and St Andrews in New Brunswick (Winick, 2021).



George Cormer, captain of a whaling vessel, made recordings of speech and song of many First Nation communities of the Northwestern Hudson Bay region. (Bain News Service, publisher – Library of Congress Catalog)

Between 1901–1902, as part of the Jesup North Pacific Expedition, Russian anthropologists Waldemar Bogoras and Waldemar Jochelson recorded 136 cylinders of songs in various locations in Siberia (Archivegrid, 2014). George Comer, the captain of a whaling vessel, was given a phonograph by the American Museum of Natural History and between 1903–1909 made

approximately 64 recordings of speech and song of many First Nation communities of the Northwestern Hudson Bay region, with the Aivilingmiut and Qaernermiut being best represented (Ross, 1984a pp. 41 & 73 and Ross 1984b, pp. 291–292).

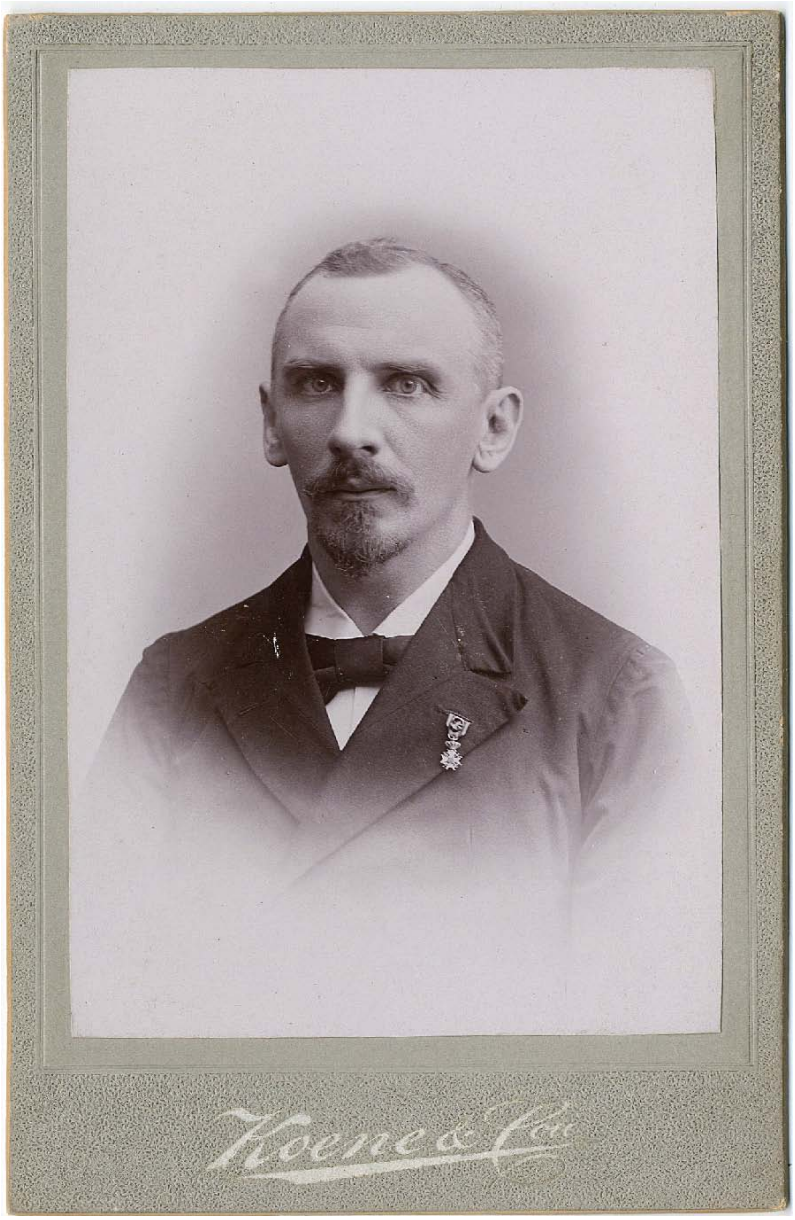


Alice Cunningham Fletcher. (Public domain, via Wikimedia Commons)



Frances Densmore. (Harris & Ewing, photographer, public domain, via Wikimedia Commons)

Also around the turn of the twentieth century, ethnographic researchers in the United States such as Francis La Flesche, Alice Cunningham Fletcher, Edward S. Curtis and Frances Densmore adopted the phonograph. The field recordings of this era, spanning thousands of wax cylinders documenting music and language, represent a priceless cultural archive with ongoing contemporary relevance.



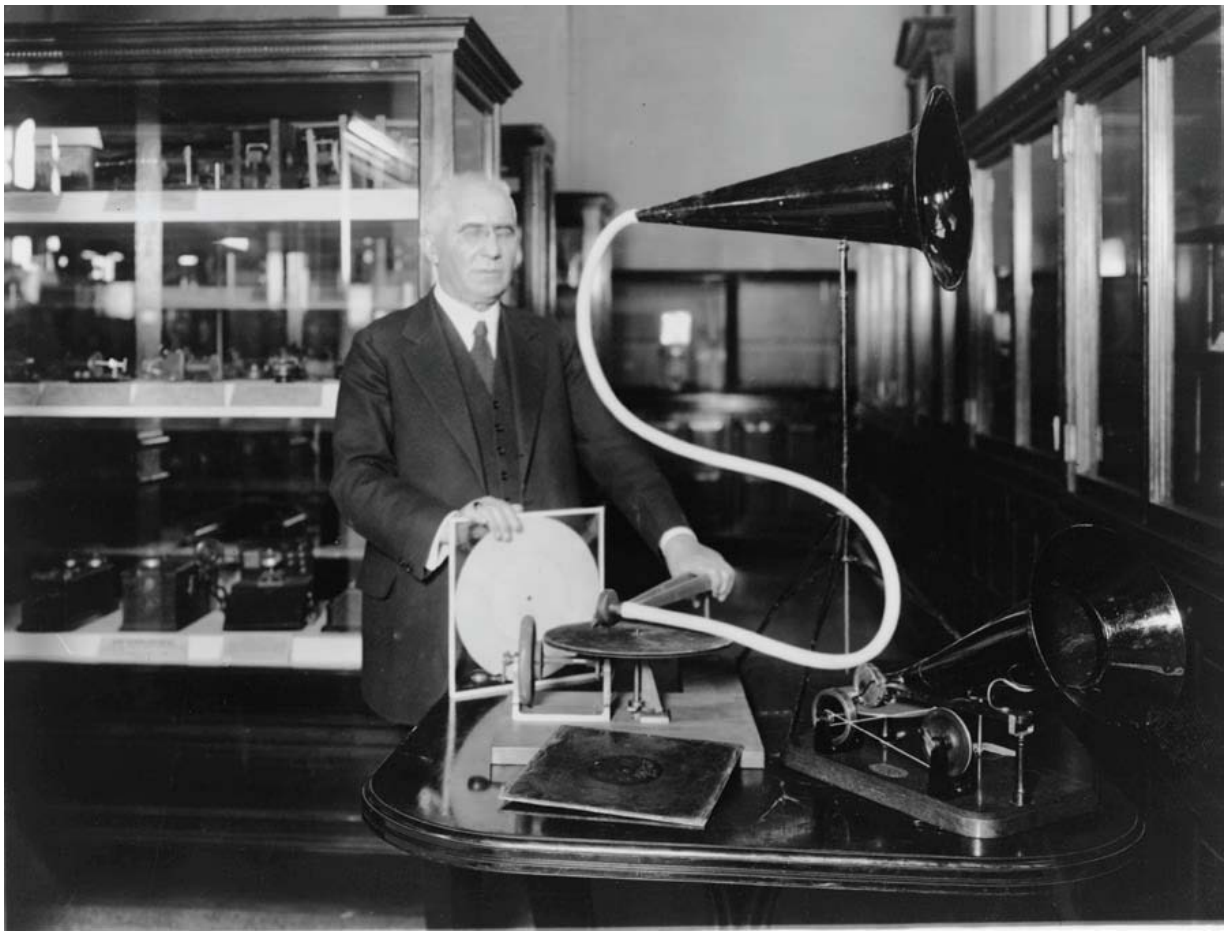
Christiaan Snouck Hurgronje made recordings of the Quran in Mecca around 1885. (Koene & Com, via Wikimedia Common)

Material from elsewhere in the world is less well documented but, for example, we have Christiaan Snouck Hurgronje's recordings of the Quran made in Mecca around 1885 (Neild, 2010) and some 20 cylinders of music recorded in Jiddah by his associate Gamal Tāg al-Dīn and sent to Hurgronje in the Netherlands between 1906–1907 (Witkam, 2007). In Hungary the composer Zoltán Kodály used a phonograph to record folk songs in 1905 and he introduced Béla

Bartók to the device, who recorded Hungarian, Slovak, Romanian and Bulgarian folk music to assist with transcription and cataloguing (Holland, 1981, p.23). In later years, he also made recordings in Moldavia, Wallachia and Algeria. Phonographs were also used in social settings: Gustave Eiffel was personally given one by Edison in 1891, which he used to record his own voice and those of the philosopher and historian Ernest Renan, astronomer Jules Janssen, author Louis Pasteur Vallery-Radot, physicist Eleuthère Mascart and his children and grandchildren (La Tour Eiffel, 2025). More generally phonographs in the home were used to record personal thoughts and musical performances.

It was not just researchers with an interest in human utterance that adopted the phonograph. Richard Lynch Garner carried out one of the first documented uses of a phonograph for animal recordings circa 1889–90 at the Chicago and Cincinnati zoological gardens as part of his ongoing study of primate language (Garner, 1892). As a child of eight in Frankfurt Am Main, Ludwig Koch was given an Edison phonograph by his father (Koch, 1956, p.15) and his 1889 recording of an Indian Schama bird is generally credited as the first recording of birdsong. As we will see shortly, Koch will go on to play a primary role in the development of both field recording practice, broadcasting and the development of audio books.

In 1898 Sylvester Judd played a recording of a brown thrasher to an audience at the American Ornithologists' Union's sixteenth congress (Judd, 1899). However, both Koch and Judd were recording caged birds. The first sound recording of a wild bird in its natural habitat was probably made by the photographer Cherry Kearton (Ranft, 2004) in the UK around the turn of the century. In 1910, Karl Reich collaborated with an engineer from Deutsche Grammophon Gesellschaft to record the song of caged nightingales and the recordings were subsequently released on Victor's Red Seal and Green Label imprints (Bruyninckx, 2018, p.61) – becoming the first commercial release of its type.



Emile Berliner, with early disk phonographs. (National Photo Company Collection, public domain, via Wikimedia Commons)

At this time, the principal limitation of the phonograph was not its audio quality; the early disk-based gramophone invented by Emile Berliner in 1886 was comparable at best and the grooves closest to the label could evidence audible speed-related distortion. At this point both systems were purely mechanical, reliant on the direct transmission of sound to the cutting stylus as captured by a horn or tube – hence only close or loud sounds were easy to record. However, disks were easier to reproduce, store and while running speed wasn't standardised until 1925 (Mudge and Hoek, 2001, p.21), a 12in disk running at 78rpm could hold around five minutes of audio as opposed to the phonograph's two minutes. Koch points out (1956, p.16) it wasn't until 1906 that he took possession of a

gramophone that could be used for good-quality outdoor field recording.

One of the earliest surviving examples (1918) of a gramophone recording made in the field is *Gas Shell Bombardment* (Adamson, 2013), made by William Gaisberg on the front line near Lille in France. This was released on HMV for the purpose of raising money for disabled veterans (it was initially intended to promote the sale of War Bonds but was not released in time). While a debate around the dubbing of some elements (such as shouted commands, bongs and whistles) in postproduction continues, the principal audio content is largely accepted (*ibid*) as authentic, being the result of the well documented field trip (Gaisberg, 1918), for which portable equipment was taken by car from England to France.

Portable is a term that needs to be applied quite loosely when talking about the first generation of mobile disk recording systems. Cutting lathes required specially prepared wax disks and these had to be preheated in an oven and kept at a constant temperature during recording – hardly ideal for fieldwork. The results could not be played back on location and they were highly fragile. Koch, for example, reports a number of breakages in transit (Koch 1956, p.27) on the way back to the studio. The advent of electric rather than mechanical recording added microphones, amplifiers, mixers and lead-acid batteries for power to the required kit. Specially adapted cars and vans became a necessity for location work. National broadcasters, record and film companies and academic institutions created appropriate resources. For example, Koch, who became the Director of a ‘cultural branch’ of the Carl Lindström Company, the makers of Odeon and Parlophone records in 1925, was presented with a ‘huge car’ filled with Neumann equipment, supplied by the German Society for the Protection of Wild Birds (*ibid*, p.26).



Technician with recording car. Finnish Broadcasting Company, circa 1939. In this era heavy equipment needed to be installed in cars and vans. (Yle Arkisto, Yle Archives, public domain, via Wikimedia Commons)

In 1924, when the concept of outside broadcasting was in its infancy, Beatrice Harrison's live cello duet with nightingales from her garden in Surrey caused a sensation when it was broadcast on the BBC on the 19th May (Kennedy, K., 2024). Public response was such that the performance was re-staged annually for 12 years. The duet was eventually recorded in 1927 and various ornithologists and bird experts have delivered conflicting opinions concerning its authenticity (*see* Dalya Alberge, 2022).

Without a means to focus in on sound at a distance (the shotgun mic was not introduced until 1956), close proximity to a source was essential for all but the loudest natural or mechanical sounds. Proximity was also essential if you wanted to isolate a particular sound from the environmental context – as Koch did, for example, during his field trips to record specific examples of birdsong. During

the late 1930s, when he was recording in England, his process would involve weeks of location hunting followed by the arrival of the 7-ton HMV recording van (which required precise levelling) and a team of up to five engineers and assistants to manage the recording process. Getting a large vehicle close to a recording location was often impossible, so running mic cables for many hundreds of yards (Koch, 1957, p.44), and in some cases over a mile, was not unusual and climbing trees to place microphones was often required. Chasing birds to drive them in the direction of the microphone became part of the practice. Multiple microphones, cabled up via a mixer, were also used, not for stereo recording (which didn't come into use for field recording until the 1960s), but to increase the chances of a bird singing near one particular mic. Playing gramophone records of bird song was also used to encourage other birds to sing.



Ludwig Koch and assistant installing a mic in a tree. Multiple mics were used in the hope that a bird would perch near one and sing. (Photo: William Deutsch, with kind permission of Anthea Kennedy)



Ludwig Koch and ornithologist Eric Hosking – more mics being installed. (Photo: Erica Koch, with kind permission of Anthea Kennedy)



Up it goes! Hoisting a mic high into a tree. Ludwig Koch looks on, second left. (Photo: William Deutsch, with kind permission of Anthea Kennedy)

Another option for location recording in the 1920s was to temporarily install equipment in a building close by. This was done at a number of locations for the *London Street Noises* project of 1928. This was an anti-traffic noise campaign run by the *Daily Mail* newspaper in collaboration with the Columbia Gramophone Company Ltd (Levack Drever, Yildirim and Cobianchi, 2021) and

represents possibly the first instance of a deliberate and purposeful recording of the urban sonic environment.

High-quality telephone lines could also be used to relay sound from a venue such as a concert hall directly to a record company studio. While this technique was certainly used for performances, it was not widespread for field recording. It was however used in Sweden in the mid-1930s when Radiotjanst AB facilitated Gunnar Lekander's recording of wild birds. These field trips had to be based near railway infrastructure where suitable telephone lines could be accessed. The signal was transmitted over the line and recorded using a steel-tape recorder at the radio station (Bruyninckx, 2019, p.194).

In the late 1920s, the sound needs of 'talking pictures' led to the development of the sound camera, a separate device to the film camera to which it could be synchronised. Here, photoelectric technology was used to convert sound to a visible, variable trace on film stock. On playback, a lamp illuminated this trace, and a second photoelectric cell generated a variable electrical current that was reproduced as sound using loudspeakers. When Fox-Case Movietone approached ornithologist Arthur Allen of Cornell University for assistance with recording birdsong in 1929 (Bruyninckx, p.61) for use as sound effects, he and graduate student Paul Kellogg found themselves with a recording truck fitted out with microphones, amplification equipment and a sound camera. The team's work in the field delivered much better results than had been achieved before. Having identified the teaching potential of the equipment, Allen and Kellogg put together their own system in collaboration with Albert R. Brand, M. Peter Keane and True McLean. By 1932 they were using the system for behavioural studies. The team published audio guide records during the 1930s, with *American Bird Songs* (1939), a collection of six 78s, representing a major undertaking.



Walter Ruttmann used early film sound technology to record street sounds, trams, machinery, crowds, music and animals to make experimental works for German radio in the late 1920s. (Unknown photographer, public domain, via Wikimedia Commons)

The German filmmaker Walter Ruttmann also used film-sound technology to record material for *Weekend* (Wochenende), an experimental work for German radio commissioned in 1928. This included street sounds, trams, machinery, crowds, music, animals and solo voices. The piece, which features sharp cuts and juxtapositions, is an early example of the type of sound collage that

Pierre Schaeffer would call *musique concrète* in the late 1940s. Optical sound was used extensively in the film industry for recording environmental sound and sound effects. For example, in 1940 Ken Cameron, the sound engineer for the GPO/Crown film unit, placed a microphone on the roof of the studio in Blackheath, London and used approximately 2000ft of optical film to record the sound of the Blitz (Legard, J., & Allen, B., 1988). Later, while working on Humphrey Jennings' film *Listen to Britain*, the 6-ton sound truck loaded with RCA optical equipment was driven all over the country, on some occasions returning with little more than the sound of a 'signal going clonk'.

By the time Cornell started publishing bird song records in the 1930s there was already an established precedent for the publication of field recordings. After two initial music and text releases, Koch, in his role at Lindström, published 11 sound books in the mid-to-late 1920s, three on animals and birds based on recordings made at Berlin Zoo (Koch, 1956, p.25). He documents arguments with the ornithologist Oscar Heinroth, who believed birds sang the same whether caged or in their natural surroundings. Koch was adamant that this was not the case and that captivity and proximity to other avian species often introduced an element of mimicry. He eventually proved his point to Heinroth using his own field recordings. He published two audio books in 1935: *Feathered Mastersingers* (Gefiederte Meistersänger), which featured 25 bird species; and *The Wood Resounds*.

While known best for recording birdsong, Koch also produced two sound books of army and navy soundscapes during the latter part of his time at Lindström, which was, by that time, under Nazi control (Kennedy, 2021). These featured sounds of manoeuvres, drill, military bands, a ship's engine room and sailors talking. He was also commissioned to make sound books of Cologne and Leipzig, featuring typical sounds from these cities such as street markets, train stations and factory sounds, as well as Nazi speeches. Following his 1936 lecture tour in Switzerland, Koch, who was Jewish, fled to England. Here, with the assistance of biologist Julian

Huxley and ornithologist and publisher Harry Witherby, he was able to realise another audio book, *Songs of Wild Birds.* This featured a pair of 78 disks with accompanying text by Max Nicholson. Three further releases followed: *More Songs of Wild Birds* and *Hunting by Ear* (an audio book about fox hunting with Michael Berry and D.W.E. Brock) in 1937 and *Animal Language* (recorded at Whipsnade and London Zoo) in 1938. Koch joined the European Service of the BBC in 1941 before transferring to the Home Service in 1942. Here he was employed to record sound effects. He visited factories, farms, several London destinations and recorded on London Transport.

In 1942, Koch had a busy year; he worked on an experimental composition entitled *Soundscape in Steel,* which featured many edited and modified field recordings of industrial sounds as well as extracts of musical compositions by Raymond Scott; and a soundscape composition called *Thick with Corn*, featuring the sounds of harvesting, other rural sounds and incidental music. The same year he produced a work variously referred to as *Victory V*, *Symphony in V* or *The Victory Symphony*. Based on the phrase at the beginning of Beethoven's 5th Symphony (the then signature tune of the BBC's European section), with every sound incorporated reflecting the rhythm or tune in some way.



Ludwig Koch with full ‘portable’ M.S.S. recording setup. He acquired his own equipment when BBC resources were stretched by the war effort. (Photo: Eric Hosking credit to www.erichoskingtrust.com)



Close up – Koch recording in the field: removing the swarf created by the cutting channel as it cut a physical groove into an acetate disk. (Photo: Erica Koch, with kind permission of Anthea Kennedy)

In 1943 he left the BBC's direct employment to take up a series of freelance contracts for its *Country Magazine* programme to develop 10 and 15-minute bird programmes (Koch, 1956, p.71). This transition occurred at a time when the BBC's resources were stretched by its war effort and he didn't have good access to recording vehicles. As a result, he acquired a disk recording system produced by the M.S.S. Recording Company. In the early 1930s the company founder, Cecil Watts, had developed an aluminium disk coated in nitrocellulose lacquer as an alternative to wax disks (Brownlee, n.d.). M.S.S. also produced high-quality cutting channels, which were the first to be adopted by the BBC (Wilmut, n.d. p.4). While wax recording remained the higher-quality standard, acetates (as they were known) provided many benefits. The equipment was

comparatively portable; a cutting channel could be built into a case similar to a large suitcase. Additional units for a speaker and amplifier were required. Acetates did not require pre-heating and were far more robust in transit than wax disks. A major benefit was that the recordings could be played back on site. Koch suggests the technology ushered in a new and important era for field recording. (The image of Koch with the M.S.S. recording setup has been reproduced with the kind permission of the Eric Hosking Trust, www.erichoskingtrust.com.)

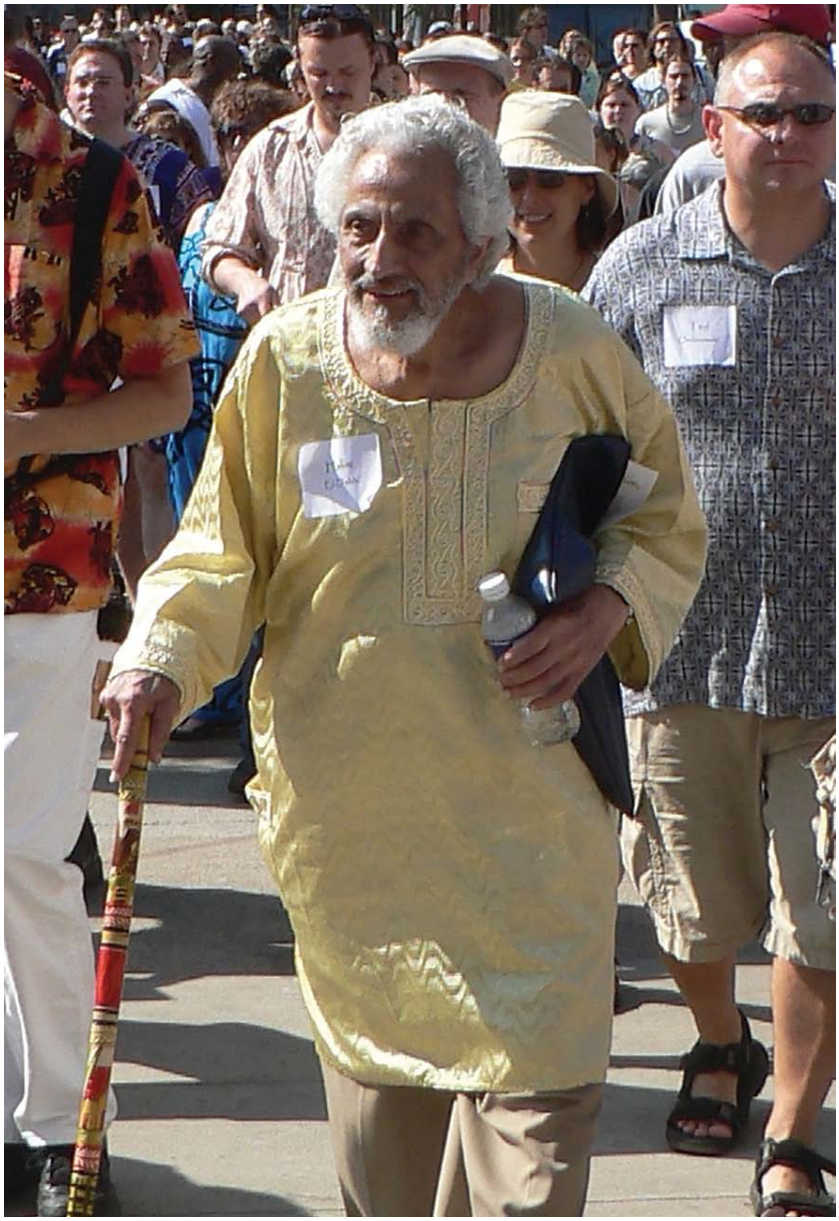
In 1945 Koch produced a soundscape for a programme about machines entitled *Masters or Slaves*. This was recorded in Birmingham and features a section where Koch had a microphone installed under a tram. In 1948 the BBC purchased Koch's personal archive of recordings (several thousand items) and hired him to catalogue the collection as the basis of its own natural history sound collection within the sound library it had been developing since 1941 (Bruyninckx, 2019, p.199). He was further re-employed by the BBC in 1949 but retired two years later, subsequently completing two more compositions with field recordings: *Music of the Sea* (1951 – described by him when broadcast as a 'sound picture without words'); *Sounds of Paris* (1952) and the audio book *Les Oiseaux Chanteurs de Laeken* (1952).



Milman Parry also used acetate disks during his visits to Yugoslavia between 1933 and 1935 to record folk songs and poetry. (Oakland Tech Centennial, public domain, via Wikimedia Commons)

Koch was not, however, the first person to adopt acetate disk technology for fieldwork: the anthropologist Melville Jacobs moved from cylinders to disks and all of the field recordings made by John A. Lomax and his son Alan after 1933 were made using a direct-to-disk machine installed in the boot of a Ford sedan (including recordings of Huddie Ledbetter, aka Lead Belly) before a transition to

magnetic tape recording around 1946. Likewise, Milman Parry used acetate disks during his visits to Yugoslavia between 1933 and 1935, to record epic songs and oral tradition poetry.



Halim El-Dabh (pictured here 2009) used a wire recorder to record Cairo street sounds, spaces and a Zaar ceremony to compose *The Expression of Zaar* in 1940. (By Halim El-Dabh, via Wikimedia Commons)

Prior to the development of the tape recorder in the mid-1930s in Germany, there were a number of moving media magnetic formats in use. The wire recorder was a practical, portable device – the concept having existed since the 1890s when Danish engineer Valdemar Poulsen demonstrated magnetic recording with his telegraphone. Wire recorders used a reel of hair-thin metal wire and sound quality was quite limited, but adequate for dictation, which was their principal function. Lower-cost domestic units were also introduced which people used for ‘talking letters’ and to record radio through the 1940s. There are some examples of them being used for field recording. When Halim El-Dabh was a student in Cairo in 1940 he used a wire recorder to record street sounds, reverberant spaces and a Zaar ceremony. These recordings were later used to compose *The Expression of Zaar*, an early example of a composition using ‘real world’ sounds that prefigures the work of Pierre Schaeffer by some years.

Wire recorders were used by US Marine Corps combat correspondents to document conflicts during WW2. For example, the recordings of Fred Welker and Keene Hepburn during the Battle of the Marshall Islands, Namur in February 1944 (Fishman & Harris, 2014). The recording captures not just the reporter’s commentary but also the sound of gunfire, shelling, troop movements and voices.



Magnetophon Tonschreiber B (1942). The first tape recorder, the Magnetophon K1, was demonstrated at the Berlin Radio Show in 1935. (By George Shuklin, via Wikimedia Commons)

While the transition to magnetic tape offered significant improvements in sound quality, recording duration, ease of playback and editing, it did not initially deliver greater portability. However, this new generation of tape machines found their way into location vehicles. Pierre Schaeffer took the Radiodiffusion Française van to the engine sheds at the Gare des Batignolles to record material for his work *Etude aux chemins de fer* in 1948 (Patrick, 2016).

During the immediate post-war years, film, TV and radio industries needed recording equipment that could be carried and operated by one person. Before the arrival of fully portable machines, the broadcaster and folklorist Tony Schwartz modified

recording equipment and installed it in his car so that he could take it onto the streets of New York to record interviews, ambiences and street music (Peters, 1985). Then, circa 1950, he acquired an Amplifier Corp. of America Magnemite recorder. This was a briefcase-sized unit designed specifically for recording in the field with an innovative design that combined spring motor drive with battery-powered electronics. Still not entirely portable, he modified it by adding a strap, moved the VU meter to the outside of the case and added connecting rods for the on/off switch, a volume control and an external mic socket.

He used this machine to record material that can be heard on his *Sounds of My City* (1959) record, which was first presented as a radio programme in 1956, and subsequently won first prize at the World Radio Festival in Rimini, Italy that year. This blends Schwartz's sparse commentary with recordings that contrast the sounds of different streets, the port, subway, bus terminal, diverse voices and interviews and live music performed by American, French, Romanian, Nigerian, Israeli, Spanish, Italian, Porto Rican and English residents. Shortly after, Schwartz purchased a clockwork/battery powered Nagra II, a much smaller, lighter unit. Launched in 1952, Schwartz was confident that his was the first in America (*ibid*).



EMI L2 Midget tape recorder. Portable recording equipment started to get smaller from the early 1960s. (By Badseed, via Wikimedia Commons)



Uher Report Monitor (later stereo model pictured here). As used by many field recordists during the 1960s, 70s and 80s. (Zlois [Diskussion], via Wikimedia Commons)



Stellavox SM4, an early transistorised tape recorder. (The original uploader was Goldfinger at Serbian Wikipedia)



Nagra III tape machine. A fully electric machine launched in 1957. (The original uploader was Goldfinger at Serbian Wikipedia)

The EMI L2 Midget tape recorder, a collaboration between EMI and the BBC in the UK, was a professional machine introduced in 1953 (Science Museum Group, n.d.). This was an all-valve unit and the weight (7.2kg) was dramatically reduced in 1960 with the introduction of the RE312, a transistorised model. Footage of David Attenborough from 1961 shows him using an RE312 to capture the first audio recordings of lemurs in Madagascar. Other innovative devices were the Swiss-made Stellavox SM4 and English Fi-Cord 1A, both transistorised units introduced in the late 1950s. The Fi-cord weighed only 1.7kg (Vintage Technics. RU, 2020) and was powered by rechargeable lead acid batteries. Nagra launched its first fully electric machine, the Nagra III in 1957. Uher launched the Report Monitor series in 1961. These robust and portable machines became a ubiquitous tool for radio journalists, and at a fraction of the cost of a Nagra, were within reach of semi-professional and dedicated amateur field recordists.



Stefan Kudelski with the spring-driven Nagra tape recorder of his own design with which he won first prize in the inaugural *Concours International du Meilleur Enregistrement Sonore* in 1952. (Erling Mandelmann, via Wikimedia Commons)

During the 1950s and 60s the tape recorder also became a consumer product. Phonography (a term dating back to the earliest days of sound recording) was starting to become a cultural pursuit in the same way as photography. In 1948 the French radio producer Jean Thévenot received so many responses to a call for contributions to a programme on sound hunting he was able to programme a series (Masson, 2022, p.4). Two years later, René Monnat produced a similar programme in Switzerland, and he and Thévenot subsequently established the CIMES: *Concours International du Meilleur Enregistrement Sonore* (International Contest for the Best Sound Recording) in 1952. In 1956 these were organised under the banner *Fédération Internationale des Chasseurs de Son* (International Federation of Sound Hunters).

In the UK, the Amateur Tape Recording Society was formed in 1955 and by 1962 there were 145 ATR affiliated clubs (Masson, 2024) around the country. Books, magazines and radio programmes proliferated to serve this substantial special interest group.



Portable consumer model tape recorder by Phonotrix (1959). (By Joe Haupt from USA – Vintage Phono Trix Portable Reel-To-Reel Transistorised Tape Recorder, Made in West Germany, via Wikimedia Commons)

Histories are generally easier to follow when the story is one of ground-breaking inventions, pioneers and famous early adopters. However, there is both anecdotal and written evidence from magazines of amateur recordists' interests. According to John

Levack Drever, his father, Ian Drever, purposefully took his tape recorder to capture the sounds of the last tram in Glasgow and it went with him during National Service in Hong Kong. Between 1957–59 a group of volunteers led by G.M. Carson produced recorded studies of life in four Derbyshire villages (Chandler, 1959). In 1964 Ray King, a salesman by profession, won both the British Amateur Tape Recording Contest and the CIMES Grand Prix in Europe for his recordings of sinks and humorous, connoisseur-like commentary on plumbing in the South West of England (Masson, 2024 and www.soundhunters.com).

Following the success of its wartime nature sound broadcasts, the BBC began to develop the range and scope of natural history broadcasting (Bruyninckx, 2019, p.201). Having purchased Koch's personal archive of recordings in 1948, the growing wildlife sound collection was taken on by Eric Simms in 1950 and the corporation set out to actively identify potential contributors. For example, John Kirby, a member of the British Trust for Ornithology (Bruyninckx, 2019, p.205). It paid licensing fees to amateur and freelance wildlife recordists, which helped to formalise relationships and fund equipment and field trips. Simms, an ornithologist and wildlife sound recordist himself, championed new recording tools such as parabolic mics and was an early adopter of tape recording at a time when disks were still the principal recording medium. He also facilitated access to the collection for other interest groups in the scientific community and duplicates of the library were provided to the Edward Grey Institute of Field Ornithology at Oxford and the Cambridge Department of Zoology (*ibid.*). Material was also made publicly available through the production of individual gramophone records and other publications such as *Witherby's Sound Guide to British Birds* (a collaboration with Myles North). This first came out in 1958 as a set of seven × 10in, 78rpm disks featuring 194 species and an accompanying text. In the 1960s, the format was updated to two × 33⅓rpm disks.

In 1964 the BBC's Natural History Unit sponsored an amateur wildlife sound recording competition (Margoschis, 1974/2018). This

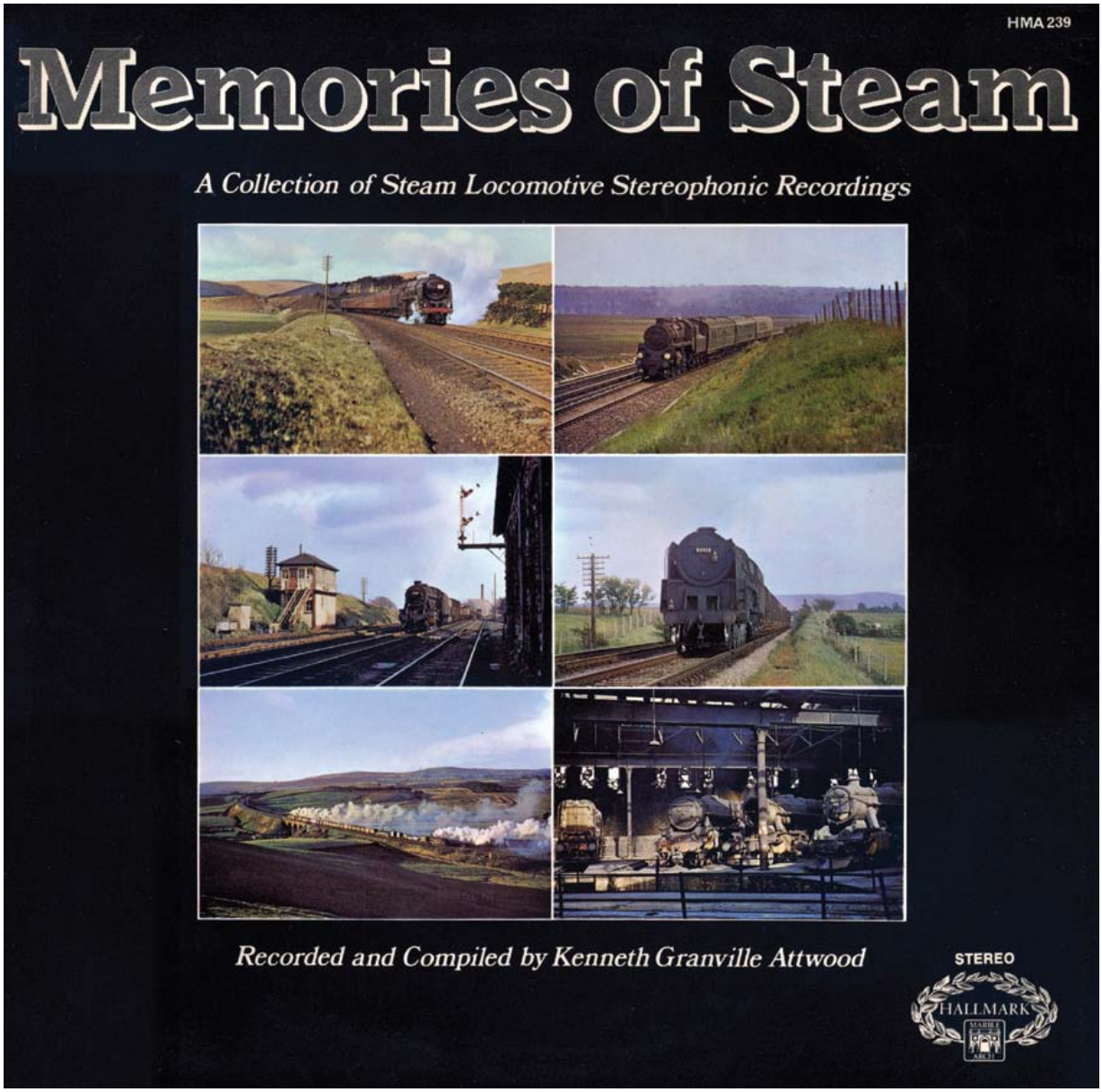
was followed by a weekend recording course at Bristol University and a weekend event at Woodchester Park Field Centre. The enthusiasm was such that the idea of a membership organisation was proposed and the Wildlife Sound Recording Society (WSRS) was founded in 1968. Strong ties with the BBC endured. The BBC's recording competition that sparked the initial interest became an annual event run by the tape manufacturer 3M (Scotch), between 1968 and 1974. The BBC further organised and ran an international wildlife sound recording competition in 1970. This also included a series of workshops for entrants, which further brought the community together and enabled the corporation to build its network of contributors (Bruyninckx, 2019, p.205). Combined action by the BBC and the WSRS encouraged the British Institute of Recorded Sound (BIRS – established in 1957) to add animal recordings to its archive (Fischer, 1957) and subsequently the entire BBC natural history sound collection was duplicated to become the basis of a new wildlife sound department within the BIRS in 1969. WSRS amateur members remained ongoing contributors to the archive, which eventually became part of the British Library. Today, the British Library of Wildlife Sounds has in excess of 300,000 catalogued and publicly accessible recordings (Trigg, 2023).

During the 1960s and 70s, gramophone records featuring field recordings entered many companies' catalogues. BBC Radio Enterprises was first conceived as a commercial venture to sell records to the public in 1965 (*see* BBC Records Blog, n.d.) and by 1970 approximately 70 disks had been released. The Wildlife Series launched in 1969 with *A Salute to Ludwig Koch*, and continued with many titles such as *Wildlife in Danger* and *Sounds of the Countryside*, with many of the recordings made by Eric Simms. Other titles such as *A British Sampler* (1969) offered a 'collection of British sound pictures for foreign visitors'. The contents of *Railways In Aspic (preserved railways as chronicled by the BBC)*; *Narrow Boats: Voices, Sound & Songs Of The Canals*; *Powered Flight* and *Traction Engines* are all self-explanatory. Other companies, such as Hallmark/Marble Arch released the excellent *Memories of Steam*

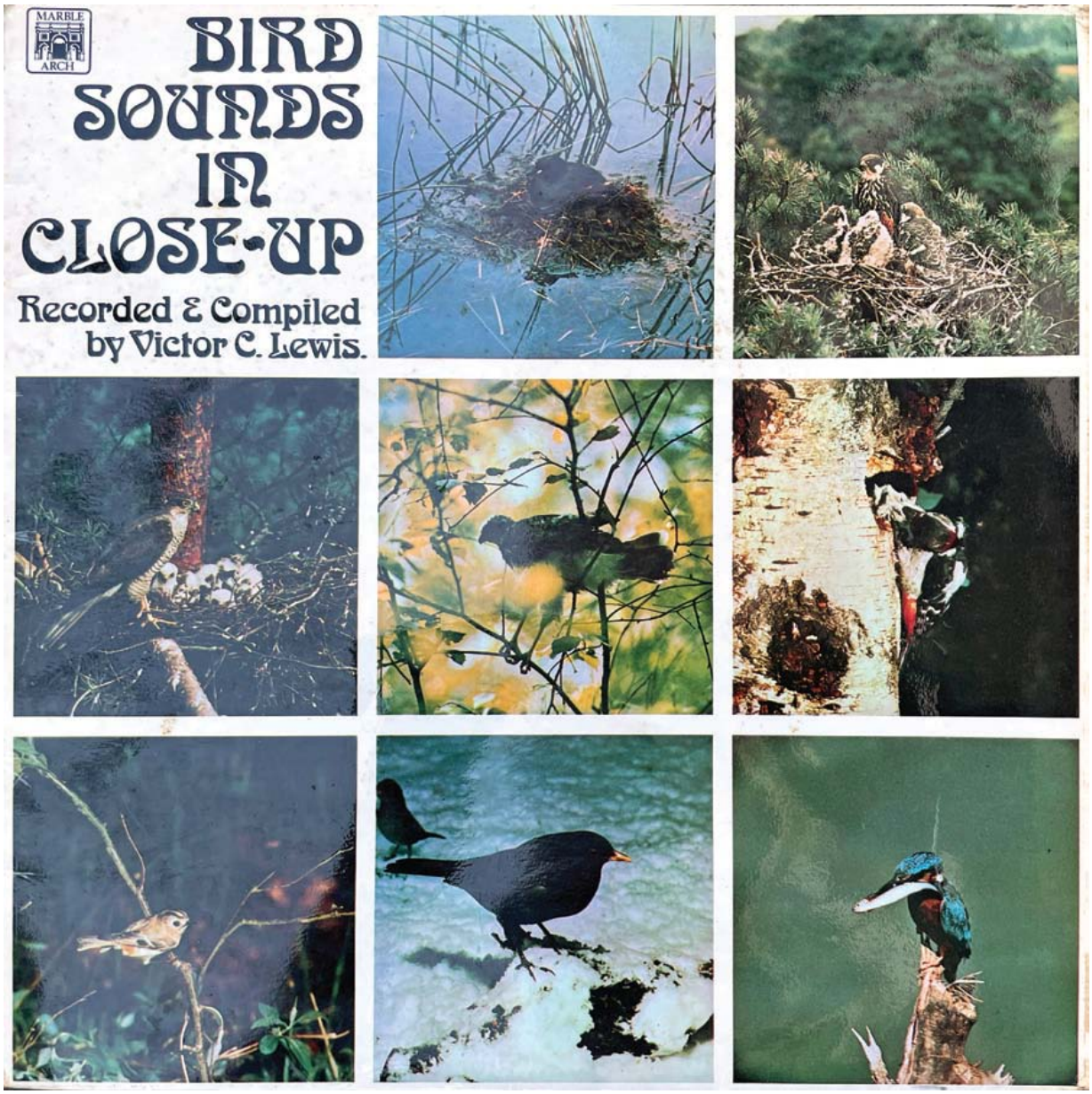
recorded by K.G. Attwood on its Pickwick imprint. This one is unusual in as much as the sleeve notes offer not only detailed contextual information but also technical information about the equipment used: a stereo Uher 4200 report monitor with either a pair of Sennheiser 421 mics or AKG D224s.

And the BBC did not have a monopoly on releasing bird recordings, as evidenced by Pickwick's *Bird Sounds Close Up* by Victor Lewis. This thoughtfully conceived release provides a mix of contextual field recordings to establish the listener's awareness of the overall location followed by close-ups of the individual singers. There is an accompanying booklet and the detailed sleeve notes stress Lewis' aesthetic which shunned 'trick mixing, superimpositions or edited sequences' and the use of 'sound concentrators' (parabolic mics) in favour of painstaking observation and microphone placement.

During the late 1960s, a general interest in environmental sound and a growing awareness of noise pollution led R. Murray Schafer, a Canadian music educator, writer and academic to establish the World Soundscape Project (WSP) research group at Simon Fraser University in Vancouver, British Columbia, Canada (SFU School of Communications, n.d.). Based at the Sonic Research Studio the group developed an interdisciplinary approach to observing and documenting the sounding aspect of environment within an overall framework that became known as Soundscape Studies. Acoustic Ecology emerged as the specific study of the effects of the acoustic environment on citizens and communities. Much of the work was done in relation to projects and published outputs, and field recording and soundscape composition became key research methods.



Memories of Steam by K. G. Attwood, 1970. One of many albums of the era featuring professional, high-quality field recordings. (Photo: Marcus Leadley)



Bird Sounds in Close Up by Victor C. Lewis, 1969. (Photo: Marcus Leadley)



Traction Engines, 1971. (Photo: Marcus Leadley)



World Soundscape Project members at SFU, 1973; left to right: R. M. Schafer, Bruce Davis, Peter Huse, Barry Truax, Howard Broomfield. (Used by permission of the World Soundscape Project, Simon Fraser University)

Unlike *Musique Concrète*, where recorded sound was used purely for its aesthetic characteristics, soundscape composition was envisaged as a way to engage the listener's awareness of content, context and origin – and enhance the relationship between audiences and environmental sound (*see* Truax, 2001). *The Vancouver Soundscape*, a two-record set with accompanying book by Schafer was released in 1973. The disks featured a mix of documentary sound recordings, sound collages and compositions. Material was recorded at many different locations: near the ocean, harbour, parks, playgrounds, a baseball game and at various community events to capture the diverse, multicultural aspect of the city.

The Vancouver Soundscape was followed by a wider Canadian recording project, which became *The Soundscapes of Canada* radio series. In 1975 a European tour was undertaken which delivered detailed studies of the soundscapes of five villages in different

countries: Sweden, Germany, Italy, France and Scotland (Schafer, 1977). The tour generated two publications and a significant body of field recordings (all made with a stereo Nagra) which, when archived with earlier material, constituted an accessible WSP library with over 300 recordings.

In terms of a legacy, Schafer's book *The Soundscape: Our Sonic Environment and the Tuning of the World*, published in 1977, is a seminal text for anyone interested in environmental sound. In this and other publications, Schafer establishes key terminology for discussing environmental sound that provides us with a shared language. Terms such as soundscape, keynote sound, soundmark, sound source, sound signal and acoustic environment have become widely used and in 2014 some were embedded within Acoustics ISO 12913-1:2014 to establish precise, international definitions. The WSP has also been instrumental in the development of a number of trajectories where field recording plays a significant role, and by extension, communities where people who enjoy field recording find a platform for their work and interests. For example, the World Forum for Acoustic Ecology, which launched in 1993, is an international network of affiliated organisations with its own journal and conferences to engage the membership.

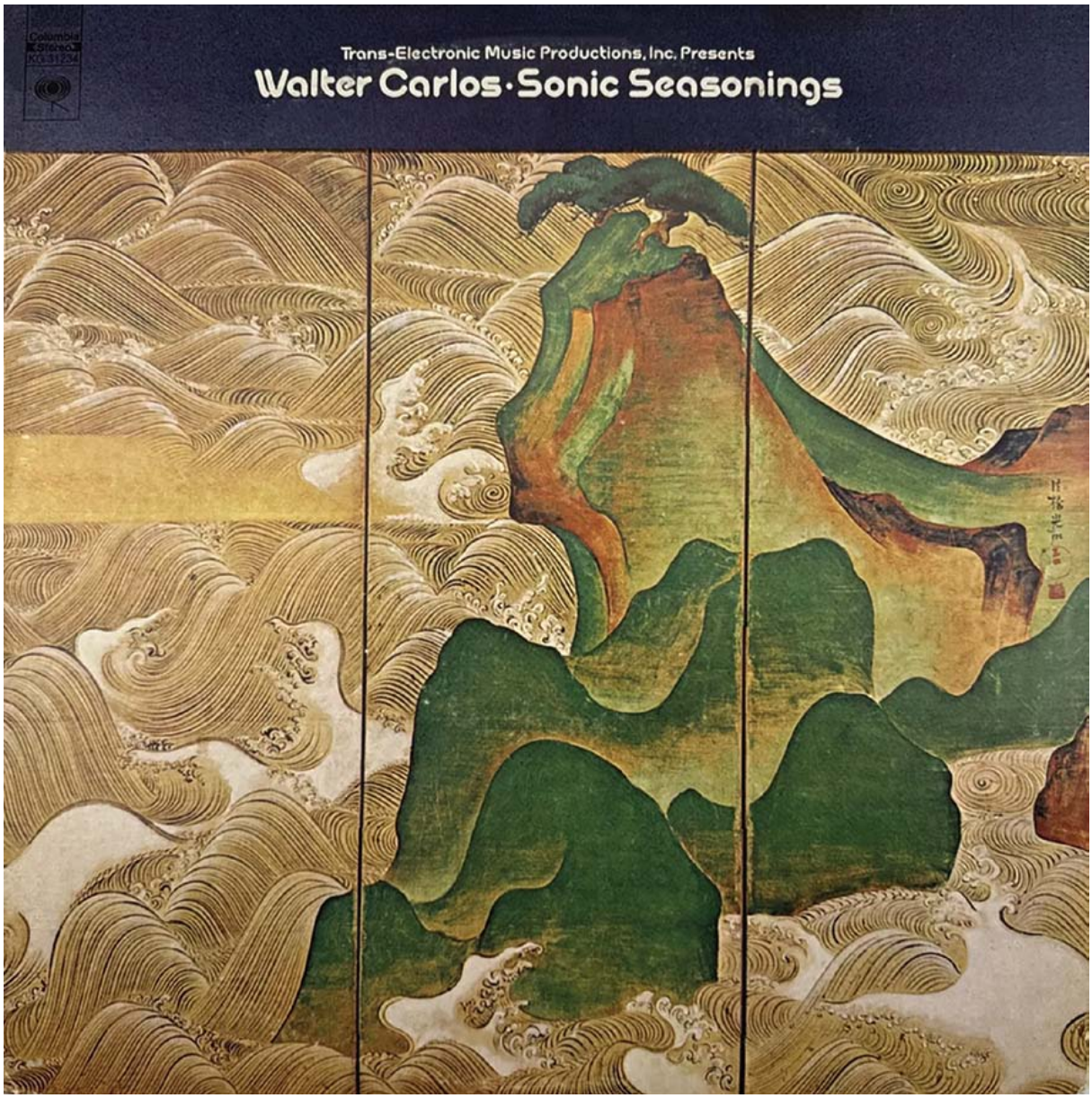


Bruce Davis and Peter Huse recording in the field. (Used by permission of the World Soundscape Project, Simon Fraser University)

While industrial, professional and academic needs continued to develop during the 1970s, the release of lower-cost, portable equipment created better access for amateur and semi-professional field recordists. Artists and musicians also began using the tools more widely. Record companies started to release an eclectic and often experimental range of music and the use of field recordings in studio production became widespread. Two albums that ushered in the decade were Bernie Krause and Paul Beaver's *In a Wild Sanctuary* and Wendy Carlos' *Sonic Seasonings*.



In a Wild Sanctuary (album artwork). One of the first albums to combine music and field recordings. (Photo: Marcus Leadley)



Sonic Seasonings (album artwork). Carlos' album is a ground-breaking work that combines synthesiser sounds and manipulated field recordings. (Photo: Marcus Leadley)

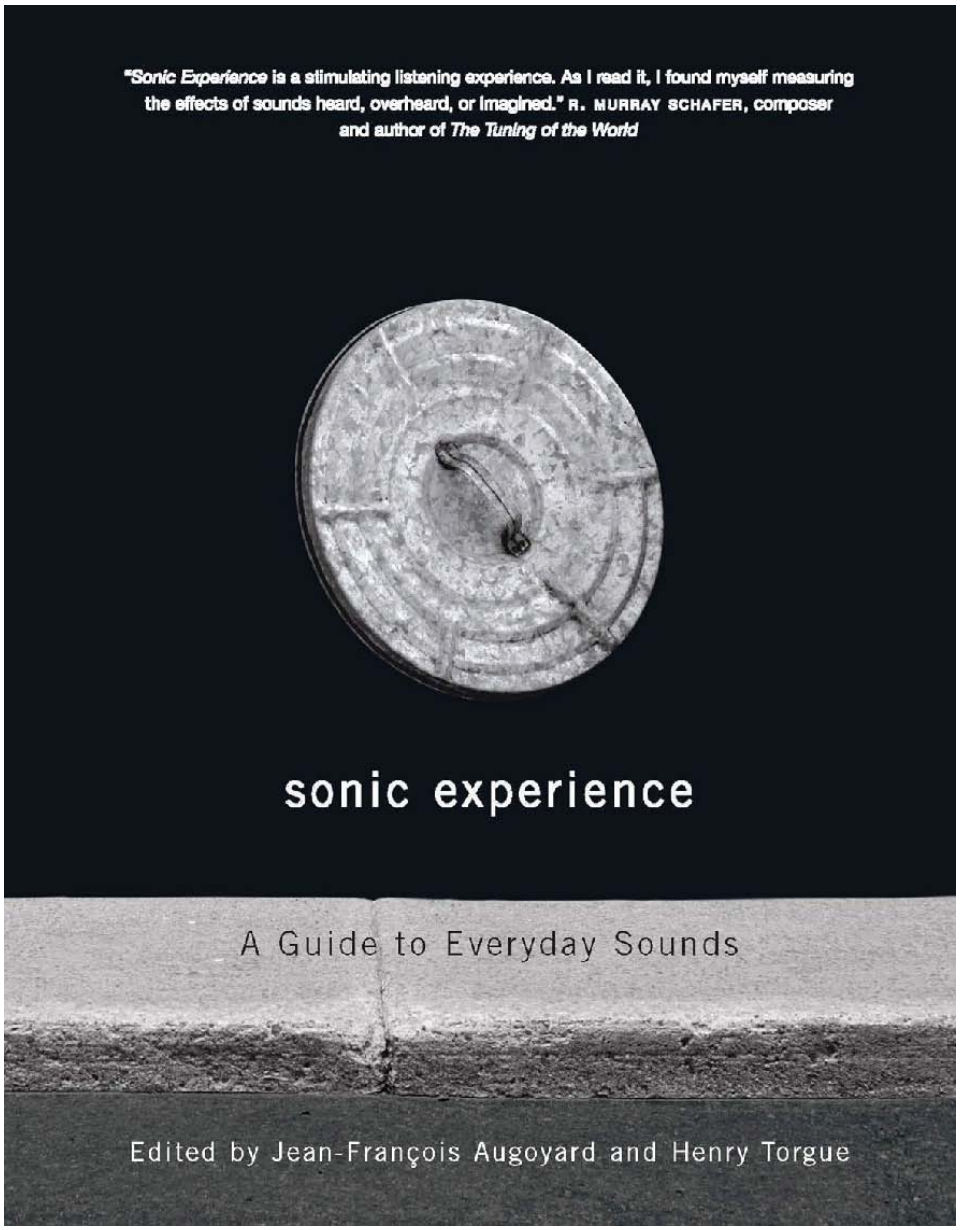


Bernie Krause. Pioneering composer, field recordist and writer. His book *In a Wild Sanctuary* is a key text for the field of ecoacoustics. (Alex Braidwood from Des Moines, USA, via Wikimedia Commons)

In a Wild Sanctuary is an eclectic mix of tracks with electronic, gospel, world and lounge music influences. The extraordinary *Walking Green Algae Blues* combines the sounds of waves, children's voices, birds, a zoo commentary, elephants and fire engines with a 12-bar blues. *Sonic Seasonings* is both a concept album based on the yearly planetary cycles and an ambient record

from a time before either genre had been formally named. At times it's subtle and beautiful with the combination of bird sound and sparse Moog synthesiser, forceful in its application of ocean waves and thunderstorms and deeply unsettling with the slowed-down sound of cows and wolves.

While Carlos went on to produce soundtracks for both Stanley Kubrick's *A Clockwork Orange* and *The Shining*, Bernie Krause took quite a different direction to become deeply involved with the field of soundscape ecology. Having established the Wild Sanctuary team in 1968 he has been travelling the world ever since recording natural environments and focusing public attention on ecological issues through publications, CD releases and sound installations. His book *In a Wild Sanctuary* is a key text for the field of ecoacoustics.



Sonic Experience: *A Guide to Everyday Sounds* offers a unique set of tools for understanding and investigating the sonic environment.

During the late 1970s, field recording became more deeply established as an investigative method in several academic disciplines. For example, the *Centre de recherche sur l'espace sonore et l'environnement urbain* (CRESSON), a multidisciplinary research unit based at the University of Grenoble, was established in 1979. Originally focused on sound and space (since the 1990s its scope has expanded to encompass all aspects of *in situ* urban

perception), the team developed a multidisciplinary approach encompassing architecture, social science and engineering methods. Field recording became an integral part of the methodology. Björn Hellström (2002, p.61) observes that study began from the position of *diagnostiquer le bien*. It was the signification of quality that was important: a space that sounded good did not necessarily need to be quiet. *Sonic Experience: A Guide to Everyday Sounds* (Augoyard and Henry Torgue, 1995/2008) provides a clear overview of aspects of CRESSON methodology and is a key point of reference for many related fields. As a guide to an eclectic range of phenomena it is grouped thematically. As a result, it bears a strong resemblance in both form and function to Barry Truax's WSP *Handbook for Acoustic Ecology*.



John Levack Drever’s practice spans music, acoustics, audiology, soundscape studies, urban planning and sound art. (Photo: Wing Man Drever)

In 1998, John Levack Drever co-founded the UK and Ireland Soundscape Community (affiliated with the World Forum for Acoustic Ecology), for whom he chaired Sound Practice (Dartington College of Art 2001), the first conference on soundscapes in the UK. Following his PhD at Dartington, he co-ordinated *Sounding Dartmoor – a participative soundscape study* (2000–2002), an environmentally orientated public-arts project exploring the soundscape of Dartmoor

through engaging with its fauna, flora and human populations. The project was supported by Arts Council England and functioned as a collaboration between the Touring Exhibition of Sound Environments (TESE), the Digital Crowd (University of Plymouth) and Aune Head Arts.



Oasis 2000 by Christina Kubisch: participants could ‘tune into’ field recordings and other sounds transmitted through overhead wires using specially designed electromagnetic headphones. (Photo: Nicola Levinsky)

During the late 1990s and early 2000s, London provided a vibrant experimental scene for artists, musicians and researchers whose practices engaged with sound and environment. *Sonic Boom: The Art of Sound*, a landmark show at the Hayward Gallery curated by David Toop, featured the work of over 30 international sound artists and was visited by nearly 40,000 people during the spring and early summer of 2000. As well as installations featuring sounding objects, gramophones and moving image content, the catalogue included two

CDs of audio recordings. Works by Christina Kubisch (*Oasis 2000*), Greyworld (*Studio 5*) and Christian Marclay (*Guitar Drag*) featured field recordings.



Peter Cusack is a field recordist and musician with a particular interest in the acoustic environment. (Photo: Misha Maltsev)

At this time both the London Musicians Collective (founded in 1975) and Sonic Arts Network (established in 1979 as the Electroacoustic Music Association of Great Britain) had extremely active event programmes and acted as hubs for information, community and collaboration. In this environment, and the wider milieu of the day, field recording, soundscape composition and acoustic ecology became valued practices and points of engagement within a broad, experimental mix, and there were many

crossovers. For example, Peter Cusack was an improvising musician and field recordist with an interest in acoustic ecology. He started his *Favourite Sounds* project in 1998 with *Your Favourite London Sounds*, which was released on CD by the LMC in 2001. The basic premise here was to ask people about the sounds they valued and record them, and the approach spread to investigations of Manchester, Chicago, Beijing, Berlin and many other cities. *Sounds from Dangerous Places* saw Cusack visit the Chernobyl exclusion zone, the oil fields of Azerbaijan and Dungeness nuclear power station among others (including later visits to the Aral Sea and Central Asia between 2013 to 2018). Recordings were released as a two CD collection with accompanying booklet. During the same period, Angus Carlyle created a 4-channel installation of motorway traffic at the Architecture Foundation (2001) and a second show at the same location featuring sounds at airshows (2003). He also produced a Resonance RM programme (with Robin Pender) called *Planes, Trains and Automobiles* (2003).



Steven Feld (one of the speakers at CRiSAP's Sound and Anthropology conference) recording in Papua New Guinea in 1982. Feld was one of the first people to develop a truly mobile recording approach: using a harness to carry his equipment strapped to his chest and hand holding mics on a stereo bar. This approach was based on a technique pioneered by filmmaker Jean Rouch to enable a recordist to follow a camera to trace the movement of sound through a space. (Photo: Shari Robertson)

In 2005, Cathy Lane and Angus Carlyle founded CRiSAP (Creative Research into Sound Arts Practice) at the University of the Arts London, focusing on sound, environment, field recording, listening methodologies and sonic memory. Peter Cusack, John Wynne and David Toop were founder members. Early initiatives included the 2006 *Sound and Anthropology* conference and the *Positive Soundscape Project* (2006–2009), a multi-university collaboration funded by the Engineering and Physical Sciences Research Council (EPSRC) to reframe environmental acoustics through a positive lens.



Felicity Ford was a presenter at *Uses and Abuses of Field Recording* symposium. Her practice is heavily engaged with the extraordinary sonic content of everyday encounters and environments. Audio is presented in the form of podcasts at her Knitsonik.com (recording daily life) website. (Photo: Mark Stanley)

This led to a final report, exhibition and commissions by Chris Watson and Yannick Dauby. CRiSAP joined the *Sounds of Europe* (2010–2013) project, culminating in *In the Field*, a British Library symposium and book exploring field recording practices. Further explorations included the 2011 *Uses and Abuses of Field Recording*

symposium and compositions for the Sound Portal (2013) at Chelsea College of Arts.



Francisco López, one of the people interviewed in *In the Field*, has an extensive creative CV including concerts, installations, projects and environmental recording. He runs field-recording workshops in the Brazilian Amazon and South Africa, among many other wilderness locations.

From 2014–2019, the *Points of Listening* programme, which became part of Salomé Voegelin's *Listening Across Disciplines* project, deepened the sonic inquiry. Related events like the *Large Objects Moving Air* and the *Audio Testimony* symposia addressed field recording's artistic, ethical and testimonial potentials. Recent work includes *Acts of Air* (2020–22), an exploration of sound interventions in urban environments and *Wild Energies* (2022), a three-day focus on and around the work of Annea Lockwood. Mark

Peter Wright's book *Listening After Nature* (2022), examined field recording through ecological, ethical and post-colonial lenses. *In The Field 2* (2024), an international conference, continued this legacy with an exploration of new perspectives in field recording practice and research.

FACTFILE 7.1: WORLD SOUNDSCAPE PROJECT TERMS AND DEFINITIONS

Soundscape: The sonic environment. The term can be used to reference any natural or built environment. It is also used in relation to compositions, both musical and those made from real-world sound and field recordings that are regarded as environments.

Acoustic environment: A combination of all the sounds that can be heard in a specific location. Schafer makes a point here that it also includes the listener's perception and 'how that environment is understood by those living within it'.

Sound signal: This is any sound to which the listener's attention is drawn.

Keynote sounds: This is the sound or sounds that form the background to the listening experience. For example, the sound of traffic, wind in trees or factory processes. All other sound signals are perceived in relation to this.

Soundmark: Shafer based this term on the word 'landmark'. Soundmarks are sounds with unique characteristics that people, especially those living within the frame established by a particular soundscape, observe as unique or of particular interest. For example, the sound of church bells.

Acoustic space: The area over which a particular sound can be heard before it is masked by the background sound.

Schizophonia: A word based on the term 'schizophrenia' used by Schafer to identify incidents of recorded or broadcast sound appearing in the soundscape. He disliked the idea of the separation of sound from the source that generated it and 'the aberrational effect of this twentieth-century development'. Personally I prefer Heikki Uimonen's term 'transphonia', which he defines broadly to cover, 'mechanical, electroacoustical or digital recording, reproduction and relocating of sounds'

(Uimonen, 2005, p.63) as it is not implicitly linked to a value judgment.

Acoustic ecology: The study of the ways human and non-human life forms respond physically and emotionally, and evidence-specific behavioural patterns that relate to the experience of their acoustic environment.

Acoustic design: A practice of engaging with a prevailing soundscape to 'improve' its characteristics. Again, Schafer's concept of acoustic design is based in a series of value judgments concerning the nature of 'good' and 'bad' soundscapes. He proposes that scientists, social scientists, artists and musicians should define the parameters for intervention. (For further information *see* Schafer 1994, pp. 271–275.)

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