Engineering in the Digital Recording Studio

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> [A graduate sound man] should be able to study a score and build up in his imagination a perfect image of the sound of this score. His ear training shall then enable him a) to find out whether or not a recording harmonizes with the image in his imagination; b) to define exactly the differences between his image and the sound of the recording; c) to indicate what should be corrected and how; d) if necessary suggest or even carry out such mechanical improvements of all acoustic, physical or mechanical devices as might be necessary for the case in question. [Schoenberg, 1940, cited in Feisst, 1999]

Abstract

The change from analogue to digital recording and mixing technology over the last thirty years has made some competencies and knowledge obsolete whilst making new demands on the skill set and understanding of the sound recordist and mixer. The use of the term 'engineer' to describe those who make and mix recordings is controversial to some since the work that is done in many studios can seem to have little to do with the 'discipline of engineering'. Has the rise of digital technology made acts of engineering within the recording studio more scarce or have they become more commonplace? Has the engineer, who once designed and maintained complex analogue audio systems via the application of scientific principles and numerical analysis, been made redundant by 'easy to use' recording and mixing systems? Or does the abundance of general-purpose processing and the quality of conversion between the analogue and digital domains, mean that engineers of software can now create, almost as quickly as they require them, the bespoke tools that they need to continue to make the very best of the recording scenarios they find themselves in and the hardware they have to hand? As part of the wider *Is Recording Engineering?* project, this paper discusses examples of procedures and practices in both the analogue and digital recording studio and the extent to which they embody common notions of the discipline of engineering and the extent to which recording has been changed by digital technology from Schoenberg's vision of trained 'sound men' and more traditional ideas of the 'recording engineer'.

Introduction

> In the field of sound recording ... we can look forward to a period of consolidation, a digestion of all that has been created in the last century. Microphones, loudspeakers, amplifiers, recorders and transmission channels are all solved problems; all the tests and ordinary experience confirm this. Now we can begin to relax and fine tune, as people have in other lines of endeavour in which the major problems were solved long ago [Dolby, 1998]

Ray Dolby, someone who might lay claim to have done more than any other individual to have extended the life of magnetic tape as a storage medium for analogues of audio waveforms, wrote those words in an article which sought to update predictions about the future of sound recording made by Marvin Camras. Camras' predictions are now fifty years old and were remarkably prescient (although he was a few orders of magnitude over-optimistic about the storage capacity of magnetic devices for digital data) [Camras, 1962]. Whilst Dolby was discussing audio technology and a magneto-electrical engineer who contributed anormously to advances in this field, his observations also beg the question about those who actually make recordings, rather than recording technology: is making a recording still an engineering challenge in the digital age? Was it ever an engineering challenge or has it always been an accomplishment of craftspeople and technicians? Has the task of recording become easier as a result of the so-called 'digital revolution'[1] or has it become more complex and demanding as a result of the new possibilities that representing and manipulating sound as finite precision numbers offers us?

This paper examines these questions by first identifying who the recording engineer is, and what they do. Then the technologies that are required and have been available for their work, both before and after the digital revolution, are compared. In this comparison of technologies particular attention is paid to extent to which they require or facilitate interventions in their operation which might typically be considered to require engineering in some form. The questions that this paper examines are, of course, part of a larger set of interdependent enquiries covering a much wider area. For example, it is an emerging and then growing cost advantage in digital technology (as well as, in many cases, functional and quality advantages) which has led to its widespread adoption. The extent of this cost advantage has significantly widened the situations and environments in which certain kinds of recording can take place and this, in turn, has had an impact on the nature of employment as a recording engineer along with its status. This paper does not address these issues: the focus is on what digital technology means for the skill set and competencies associated with 'recording engineering'. The intention is to identify where the discipline of engineering is now found in recording situations as a result of the move from analogue to digital technology.

Who is the recording engineer?

You do not have to search far to find controversy relating to the use of term 'engineer' in relation to recording. The following is an abridged version of an exchange about the *Wikipedia* entry for 'Audio Engineering':

> **Unsigned entry:** An Engineer is one who has a degree in engineering, and in my opinion, those licensed to practice engineering ... Audio technicians (what this article refers to as audio "engineering") operate the equipment and mix the sound. While

[1] The digital revolution refers here to the rapid and almost all-pervasive adoption of audio technologies which use finite length codes, rather than direct physical analogies which are continuous, to represent sound. The Compact Disc, a consumer medium, was introduced in 1982, professional digital recording and editing systems pre-date this although analogue recording remained prevalent and economic until well in to the 1990s.

> that is definitely a skilled trade or craft, it is not engineering. Engineers DESIGN the equipment that is used in those fields. Designing the equipment from the transistor and discrete electronic component requires knowledge and training far and above operating the finished product.
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> **PhilyG:** ... I'm assuming you have done absolutely no work in the music industry, correct? Look on every single CD, cassette, or LP that you have in your house, the people who recorded and mixed the album are credited as engineers. Everyone who works in this industry fully understands the difference between a creative audio engineer who has creative mix and recording input and a designer who is skilled in science.
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> **Unsigned entry:** ... I work in the telecommunications industry. I understand they are credited as "Engineers" on record labels. I don't care that this may have been done by the recording and entertainment industries for 60 years, it is still a misapplication of the title to people not qualified as engineers. The audio "engineers" you describe are generally not scientifically trained. They are still technicians in the realm of engineering which again is a learned profession ... Operating the sound equipment and maintaining it are both technical and skilled trades but they are not engineering .. Record companies can call their employees anything they want (unfortunately) but engineer is something that one becomes by virtue of engineering school and licensure regardless of what an industry may call it.
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> **PhilyG:** I'm totally okay with a making a note that Audio Engineers are not certified [professional engineers], but I would hate to say that we are really "technicians" who have been mistakenly labelled by the music industry for years. We engineer the music that comes out of the speakers just as you engineer the electricity that flows through devices. We are not simple operators of the equipment, it is a creative field.
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> **Unsigned entry:** You do not design anything. You are not creating new equipment and technologies. Yes you do a creative job, but it is still not engineering. [unattributed, 2011]

This exchange epitomises one of the issues with the combination of 'recording' and 'engineering'. One view of engineering is that it is something which is only achieved by formally trained and then chartered individuals who use quantative methods to solve problems and achieve outcomes. The alternative view expressed above in relation to recording, is that it is the application of some kind of skill in using technology in order to produce an artefact – the pressure waves generated by the performance are captured and martialled by the recordist so that they can be recreated (or at least re-evoked) for the listener; the recordist, in actually making the recording, solves the 'problem' of how it should be done. This immediately raises the problem of evaluating the quality of the outcome. In many traditional engineering[2] situations it is possible to objectively measure the quality of the outcome. The quality of a recording is, certainly to some extent at least, subjective: as one microphone designer observes "there is no adequate basis in pure physics for judging the sound of a recording" [Wuttke, 1999]. That said, it

[2] It is acknowledged that, given the subject matter of this article, the term 'traditional engineering' could be considered a controversial and ill-defined one. Here it used to work undertaken largely by workers who have successfully met the criteria for some form of formally recognised engineering charter.

will usually conform to some broad requirements. For example, only in rare circumstances would a recording for audio CD in which both left and right channels are identical (i.e. it is in mono rather than stereo) be desirable or acceptable. This is quantifiable as a requirement for a "non-zero energy difference signal" and can be checked with tools such as a phase scope (Lissajous display), but can also be evaluated by listening with two ears. Of course recordings exist only to be heard[3]; they will not be driven over like bridges, expected to start on cold mornings like cars, required to perform the myriad different tasks of a smart phone or personal computer and so perhaps quantification of them is completely unnecessary. David Blockley makes a similar comparison:

> Many years ago I was listening to a well know jazz critic interviewing Louis Armstrong on the radio. After a long erudite discussion of the contrapuntal complexities of his trumpet playing the critic asked 'Louis how do you do it?' The reply delivered in typical rasping style was, 'Man I just blows.' Why is this relevant to engineering? Because, like Louis, much of what practitioners actually do is as a result of experience - they learn from doing the job. However if Louis had blasted a wrong note then his career would have suffered - but little else. The consequences of engineers or indeed many professional practitioners such as medics, making a wrong decision can be loss of life [Blockley, 2008].

Here it is the criticality of the task, its potential to determine human life, which is contrasted. It is perhaps a limited point since many applications of engineering, the design and implementation of the graphics engine for a computer game for example, do not determine human life. Although Blockley does not appear to be claiming that engineering *must* involve critical activities whose outcomes determine life or death, the reliance of modern civilisation on engineering is often cited when defining the importance of this discipline [e.g. Willis et al, 2009].

In the 1980s Douglas Lewin identified 'Two Cultures' of Arts and Sciences which dominated universities and proposed Engineering as a third culture. In what is partly a manifesto for the appreciation of engineering as a discipline separate, rather than extending, from science he observes:

> Though today engineering artifacts draw extensively on scientific knowledge, it has not always been so, and up to the end of the nineteenth century the methods of manufacture and natural science were quite distinct. Indeed the development of the steam engine, often quoted as a 'product of science', was the work of men like James Watt and George Stephenson who were predominantly skilled craftsmen and certainly not knowledgeable in scientific matters. Thus engineering can be said to predate science, which is still relatively young, and man's development has been, *and still is*, determined essentially by his capacity to make artifacts and improve upon his environment rather than the systematic accumulation of knowledge [Lewin, 1983].

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[3] It may be that, at some point in the future, audio recordings become information artefacts to which machine analysis can usefully be applied. At this point they will acquire a utility which can be enjoyed outside of human listening. However the assumption here is that recordings exist solely to be listened to.

The distinction that Lewin makes between the two is that science often (although not always) deals with systems which are ‘closed’, isolated from the real world, whereas engineering is concerned with open systems which must serve and operate within real-world situations. From this we might consider that engineering, in a recording situation, delivers ‘what is possible’ given what is known about laws of nature and the limits of technology, along with ‘what is desired’ according to aesthetic considerations. However, before we become too comfortable with the idea that we have identified a convenient slot for recording within the discipline of engineering, there is still Lewin’s delineation of the ‘designer craftsman’ and ‘designing engineer’ to consider. In the dialogue quoted previously the correspondent who was resistant to the term ‘recording engineer’ was happy to concede that the act of sound recording was a “skilled trade or craft” [unattributed, 2011]. Lewin elevates the status of engineering above this in stating that “the design of artefacts such as a communication system, computer software system ... etc. require[s] a high level of intellectual skills rather than simple craft skills” [Lewin, 1983]. Writing in the same year as Lewin, Gordon Rogers states that “craft is the power to produce a preconceived result by consciously controlled action: the craftsman always knows what he wants to make in advance” whilst the engineer will innovate and have to choose from a number of solutions to a problem [Rogers, 1983].

Writing a decade before Lewin, John Borwick described the ‘Tonmeister concept’, which has existed since the late forties in Germany, using as its basis Schoenberg’s idea of what the qualities of “trained soundmen” should be. This concept arrived in higher education in the UK at the University of Surrey in 1970. Borwick had previously worked at the BBC as a Studio Manager, a role which had previously been referred to as Programme Engineer, before becoming involved in that corporation’s training centre at Evesham and then moving onto Surrey to establish the four-year (including a year-long industry placement) Bachelor of Music (Tonmeister) degree. He noted:

> This word ‘tonmeister’ takes a bit of explaining to British and American studio personnel, though it is well enough understood in recording and broadcasting studios all over the Continent of Europe. A literal translation from the German produces ‘sound-master’, not very helpful, but at least implying that a tonmeister is someone skilled in the arts of sound recording, transmission and reproduction. The complication, and the special delight, is that it calls for a strange mixture of artistic flair and technical knowledge. How you combine the seemingly incompatible aptitudes of art and technology in a single individual and in what proportions, has always been a matter for discussion – and even of heated argument [Borwick, 1973].

Unsurprisingly for a course residing in a Department of Music he states “our view that the tonmeister is first a musician and second an engineer”, although does not make it clear how this view influenced the design of the course. His discussion seems to relate entirely to the recording of ‘art’ as opposed to ‘pop’, ‘rock’ or ‘folk’ music and the course was originally devised to “to meet the perceived needs (in terms of employee recruitment) of the British classical-music recording industry” although “the remit of the course has expanded to include all areas of the Audio Industry and the recording of all genres of music" [Fisher, 1999].

All of this begs (and hopefully begins to answer) these questions: what is the person who makes recordings of music required to do and does what they are required to do encompass the innovation and informed decision making that characterises engineering? The existence of performance venues with microphones permanently rigged in one place (often positioned with reducing visual obstruction in mind, rather than optimising sound quality), connected to recording systems with a set of step-by-step instructions to enable recording by a minimally trained person, is an indication that certainly not all recordings are engineered: in this situation no one is exercising choice or innovating, there is just the "preconceived result". However it could be argued that the overall system, as embodied in the microphone positioning, the choice of system components may have been engineered to best match a specification: a system capable of making the best recordings given that the microphones must not visually obscure performances, their position must be permanently fixed, the operator of the system will have little or no training etc. But an individual who has knowledge of acoustics, is aware of the capability of the equipment at his/her disposal and understands the musical intentions of performer(s) and producer and exploits these in adapting to the recording situation at hand is designing a recording, making choices and may be innovating. As our microphone designer observes:

> A person who understands something about the way microphones work will be well able to master the most diverse assortment of recording tasks. "Cookbook" approaches are certainly easier, and they can be in step with the latest fashion trends, but they are not much help in the constantly changing conditions of live recording. Recording is not easy; being a recording engineer is truly a profession [Wuttke, 1999].

From analogue

A channel is a path through which a signal can pass or in which it can be stored. It may be a volume of air, a length of copper cable or a groove cut into a vinyl disk. Since the signal is transferred/stored as the analogue of one, or more, of the physical characteristics of the channel, the faithfulness of this analogue to the original signal is directly related to such characteristics. As examples these characteristics may be the elasticity of the material into which the signal is etched, the coercivity of the magnetic particles whose orientation represents the signal or the resistance of the copper cable. It is also the interaction between medium and transducer that defines the channel so channel quality might also be dependent on the velocity of the stylus in the record groove or of the tape that moves past the electromagnet. As well as its physical characteristics a channel, or any system, may be described in terms of the difference between a signal input to it and the altered signal at its output.

Because of this direct analogy between a physical property of a medium and the audio signal it represents there is no mechanism for distinguishing between other causes of changes in that property, where that property does not vary in exact proportion and where the relationship between the signal and its analogue is frequency dependent. Therefore a channel, like any system, can introduce noise and non-linear distortion into a signal as well as frequency dependent changes in level and phase. It may also introduce an overall delay and change in level between its input and output. An important design concept for the majority of audio recording, transmission and reproduction equipment is that of ‘fidelity’ to the original signal. This fidelity to the original signal for analogue equipment is determined by the dynamic range, linearity and frequency response of the channels through which the signal passes.

Electroacoustic transducers are parts of the audio signal path which are still analogue in recording situations (ongoing experimentation with digital loudspeakers notwithstanding, e.g. [Busbridge et al, 2002]). The best microphones and amplifiers/loudspeakers offer almost perfect linearity in their intended operating ranges, but they certainly do not possess a flat frequency response nor precise directional characteristics. The colouration of sound offered by the non-flat frequency response is the main contributor to the ‘character’ of these transducers, something which is particularly celebrated in favourite microphones although a greater uniformity in response is usually desired from loudspeakers. It is therefore the selection and positioning of microphones relative to the performer(s) in an acoustic space which is regarded as one of the (if not *the*) key task of the recording engineer.

The analogue storage of sound within the recording studio in the last half of the previous century was entirely on magnetic tape, thanks in no small part to the development by Camras and others of high frequency biasing which dramatically improved the linearity of the medium. Despite these and many other improvements over the lifespan of magnetic tape as the all-pervasive recording medium, it suffers from a limited dynamic range (certainly far lower than that of the human auditory system for physically manageable tape widths) and variable frequency response. Dolby’s most notable contribution to analogue recording was the set of noise reductions systems that bear his name [Bubbers, 1998]. These essentially performed the same role as modern digital perceptual coders, such as mp3 and AAC+, in ‘hiding’ noise underneath the signal and they were remarkably successful: Dolby Spectral Recording allowed 2 inch multitrack analogue tape to compete with 16 bit digital systems in terms of dynamic range, if not flatness of frequency response [Dolby, 1987]. However tape machines which could be realigned (in order to retain their frequency response) with Dolby units for each separate track (which also required alignment for each change of tape) were expensive to acquire and to maintain. Whilst alignment can be considered a procedure which can be performed by a technician rather than an ‘engineer’, the approaches required throughout the recording process for cheaper, less optimum equipment make more demands of the recordist. Cheaper systems with lower tape speeds, a lack of ‘off tape’ (confidence) monitoring, narrower ‘track per tape width’ and simpler, non-alignable noise reduction systems (such as Dolby B or C) often require recording decisions based on a more profound understanding of magnetic recording, human audition and signal processing in order to use other parts of the signal path (e.g. equalisation, dynamics processing) to achieve the desired result.

The processing of sound possible with analogue-only equipment can be divided into to three broad and overlapping categories: dynamic range control (e.g. compression), spectral manipulation (e.g. equalisation) and time domain processing (e.g. delay). Although the latter can be achieved to a limited extent within the analogue domain it is in this area that digital technology first began to make its presence felt in the studio, such as via the EMT 250 reverberator [AES, 1999]. Analogue processors all function in 'real time', there is no offline or batch processing. In fact the only task that is performed offline in the analogue studio is editing: the reordering of portions of sound on the tape by cutting and then splicing. These three sets of processors are what is available to the analogue recording engineer for the creative and remedial adjustment of audio signals. For example, where there is insufficient dynamic range available in the recorder then signals can be compressed 'to tape' and/or expanded 'from tape' (in addition to whatever compansion is being automatically applied by noise reduction systems. Analogue processors function via the direct action of physical components (e.g. networks of electrical resistors and capacitors) on the representation of the signal (e.g. varying voltage). Such systems also require circuitry for audio input and output, supplying power and neither this, nor that required for the processing, can be easily shared between different processors: they are not general purpose, modular or configurable. It is rarely practical to design and construct an analogue processor for a specific recording situation. The recording engineer can use these tools within the limits of the user-controllable parameters they offer, but they cannot design, construct and then test them during the process of recording. Of course, the 'recording studio' as a whole can be considered a modular and configurable environment via its patch bay and the interchangability of many of its components.

The process of tape editing is 'destructive' – the medium can never be perfectly returned to its previous state after a cut and splice. 'Comping', the process of assembling a continuous recorded performance by 'dropping in' for different sections of that performance during different takes, is fraught with the danger of mistiming the drop in and drop out points: wanted parts of the current take are missed, or the wanted parts of previous takes are erased and replaced.

Therefore we see in the analogue studio that the challenge of obtaining and retaining sound quality can be significant, although it could be said that the presence of noise and distortions in analogue media actually act to conceal other problems. To optimise the outcome the engineer can configure the studio at a unit processor level and can control those processors via physical controls that are provided to the user by the manufacturer. They are constrained by the granularity of the tools at their disposal, albeit aided by the fact that they are designed for typical recording scenarios. The interventions the engineer can make must either happen in real-time or be destructive (both in the case of comping).

To digital

By sampling and quantising a continuous signal it is transformed into discrete data; at regular instants of an analogue signal a digital code is assigned to represent its amplitude at that instant. The bandwidth of the digital representation is determined by the frequency with which the analogue signal is sampled and its dynamic range by the number of bits used to represent amplitude. This code still requires an analogue channel for storage or transmission but, provided the bandwidth of the analogue channel is at least the capacity or bandwidth required for the digital data and the noise level is not high enough to introduce uncorrectable errors in the code, the quality of the representation is independent of the channel: non-linearities, noise and variable frequency response in the channel are *not* heard in the signal it carries. During the 1980s and 1990s this capacity requirement was significant. In 1990 the cost of magnetic disk storage was around $5 per megabyte, giving a cost of $50 for storage of one minute of CD quality stereo audio. In 2000 storage was closer to 10 cents per megabyte, requiring just one dollar to store a minute of CD quality audio [Thompson and Best, 2000]. In 2012 we are now at the point where storage can be acquired for around 15 cents per *Giga*byte (around 100 minutes of CD quality audio). Over this time the rate of data transfer to and from a magnetic disk drive has rapidly increased too. Current mass market drives can easily deliver 24 tracks of better-than-CD-quality audio, something which in the early 1990s only a purpose built multitrack tape machine could hope to deliver. However once bandwidth and capacity were matched by digital technology then the benefit of this technology, high dynamic range and flat frequency response, had an impact particularly for studios who had only previously been able to afford modest analogue systems – the performance of their recording systems now matched or exceeded that of the analogue equipment that even the top flight studios had been able to afford. With much less noise to hide when recording, it can be argued that the level of skill required of the engineer in this particular area in order to make satisfactory recordings with digital equipment is lower.

In addition to the benefits of high fidelity at low cost, the processing of audio as data “allows tremendous opportunities which were denied to analogue signals” [Watkinson, 1994]. In addition, rather than requiring a specific type of electronic device for a particular processing task, a general purpose hardware digital device such as a central processing unit (CPU) or digital signal processor (DSP) can be adapted to a different audio signal processing task if given a new set of instructions (i.e. different software is installed and executed). This adoption of general purpose computing hardware and operating systems has had a number of dramatic effects, including an erosion of a studio as the only environment containing equipment with which audio can be recorded and processed. At the same time the number of things that can done to sound in the studio, and the different environments (e.g. real-time/offline, software/hardware, visually orientated/text based) in which they can be performed. So on the one hand the engineer’s job is made easier by cheap yet very high-fidelity recording, yet the possibilities for (and quite likely therefore, the expectations of) interventions in other areas of sound quality are increased. Even then, the fact that processes themselves are represented by code rather than physical components enables 'total recall' of any recipe for manipulating sound, allowing presets to be configured by experts and then distributed to individuals who might have become relegated to the role of mere operators of technology: technological 'secretaries' to producers, or subsumed into the role of producer and no longer a distinct entity within the studio. That is not to say that there are no operational challenges in the digital studio: complex audio software is never 100% bug-free, there can be issues of compatibility and system complexity which require negotiation and often solutions to them have to be engineered. In addition the input-output latency of digital systems was a problem (and still is in some cases) and confidence monitoring is not possible. However, when operating 'out of the box' (i.e. preconfigured and ready to use at the point of purchase) and controlled by someone with a basic understanding of how 'window – icon – menu – pointing device' systems, high fidelity recordings can be made quite easily using this equipment. In addition, that well known menu item 'undo' when applied to both recording and editing, makes many operations less risky than in the analogue studio, reducing both stress and required competence.

Before we become too depressed (or excited, perhaps) that automation, perfect recall, undo-able operations and near-perfect fidelity, have banished the discipline of engineering from the recording studio we should acknowledge the opportunities that are on offer to extend the repertoire of processing tasks that can now be undertaken at the same time and place as the recording itself. It is interesting to note that in an article entitled "The Engineers Who Changed Recording", *Sound on Sound*, a UK-based popular music technology magazine cited as their five examples individuals who had designed or invented recording equipment as well as making recordings themselves (Joe Meek, Tom Dowd, Bill Puttnam, Tom Scholz and George Massenburg) [Daley, 2004]. Possible hyperbole of the popular press aside there is an intimation here that the 'all round' engineer is not only able to use the tools of the studio to make records, but has sufficient understanding of the technology themselves to be able to invent new tools which offer greater economy or quality to the tasks they undertake in their recording roles. Given the limitations in creation and configurability of analogue equipment outlined previously, this 'complete' approach to record engineering is not one that could be feasibly replicated throughout recording studios. In the digital studio however there is more opportunity for this kind of research and development to take place.

The digital audio workstation (DAW), a computer based system for recording, editing, mixing and processing audio typically offers the ability to 'host' third-party processing algorithms. The term 'plug-in' refers to a computer program that can interact with another (the host) to provide a specific function. Very often such plug-ins are dynamically linked libraries meaning that they can be installed independently of the host and so do not require the host software to be reinstalled every time a new plug-in is required. Thus a hierarchy of functionality exists: there is a host program which provides a user interface and functions such as audio playback, below this there is the plug-in which may be used to perform specific audio processing tasks and below this there are library routines which the plug-in can use to perform specific tasks such as calculating the Fast Fourier Transform of some data or to display a graphical user interface. Plug-ins must conform to a specified format in order to function correctly within a host program since both the host and the plug-in must know what data, and in what format, each requires from the other and when it is required, and/or how to request it via function calls. There are many audio plug-in 'standards'. Some are platform/OS specific such as the Apple's Audio Units for their Macintosh computers or DirectX plug-ins for Microsoft Windows. A popular cross-platform specification is the Steinberg Virtual Studio Technology (VST) plug-in which is supported by many DAW applications such as Steinberg's own Cubase, Nuendo and Wavelab and those by third parties such as Cakewalk's Sonar and Plogue's Bidule. The software development kit for VST plug-ins is provided free of charge and essentially supplies a C++ 'wrapper' into which C code describing the real-time process can be inserted [Steinberg, 1999]. A VST plugin to perform a simple processing task such as time-delay panning can be coded and compiled in under an hour. Whilst taking such lengths of time out of recording sessions in order to produce bespoke tools for a particular task is unlikely to be practical, a recording engineer can create processors which directly relate to the way in which they work and the environment in which they do that work. They are no longer merely operators, they are designing and implementing certainly using numeracy and likely science too. Some of the tools that this author has developed for research purposes, but also for his teaching and recording work can be found at www.jezwells.org.

Whilst such endeavours in coding produce genuinely bespoke tools that can be used in industry-standard systems, developing complex or computational demanding solutions to problems can still be an extremely time consuming process. Whilst the audio input/output and user interaction is handled by the host application, the sparsity of functions offered by the cores of languages such as C and C++ means that commonly encountered audio processing tasks have to be re-coded or time-consuming searches for suitable libraries have to be found. Matlab is a software environment for 'scientific computing' and it offers alternatives to the C++/plug-in approach. Algorithms can be implemented much more quickly since there is a wide range of functions within the environment for performing common mathematical tasks many of which can operate on entire vectors and matrices (as opposed to performing operations element-by-element as is the case with the core C/C++ languages) [Mathworks, 2012]. Matlab is not suitable for real-time audio processing, but has a set of functions that facilitate audio file reading and writing and playback which enable it to be used for offline processing. Because it uses a higher level 'interpreted' language it executes tasks much more slowly then pre-compiled and linked C/C++ code, but compared to the gain in development speed this can be a minor or even barely noticeable inconvenience and interfaces between Matlab and DLLs written in C exist for use speedier execution is required. Although Matlab is relatively expensive, compared to many well featured DAW software products, there are open-source alternatives such as Scilab and Octave. Again, tools that this author has implemented in Matlab for sound recording tasks can be found at www.jezwells.org.

VST plugins and Matlab both use the central processing unit of the host system to process audio data. Whilst CPUs offer generality this comes at the expense not being specialised for any particular task. Very often the way they work on audio data, when the abstractions of language are stripped away and the pure movement and of data and application of logical operations is observed, can seem absurdly laborious, perhaps sometimes akin to moving a large pile of earth with a number of teaspoons, rather than melting those spoons down to create a bucket which can be used much more efficiently. CPUs now possess such brute strength in terms of the number of operations per second that they can carry out that this is hidden from the sound recordist...until they require yet another reverberator or instance of a voice processing channel and suddenly the CPU runs out of processing capacity. As discussed earlier the production of bespoke hardware, with power supplies and audio input/output capability is not economically feasible within the studio. However so-called ‘field programmable gate arrays’ (FPGAs) do offer the illusion and, to most intents and purposes, the benefits of bespoke hardware design, along with the advantages of parallel processing [Maxfield, 2004]. In fact the hardware is memory which ‘imagines’ it is hardware by remembering the truth tables that define the hardware that it is mimicking. Parallelism, only just emerging in CPU design through ‘multi cores’ although fairly mature in hardware processing for graphics, is a very good fit for sound recording which typically requires a number of separate signals to be recorded, replayed and processed at the same time. FPGAs allow digital hardware (e.g. memory and logical/arithmetic operators) to be specified and implemented and many are supplied with development systems which include audio input and output capability – hardware designs, including ‘off the peg’ embedded processors can be developed and run for audio operations, using such systems and a PC running the design tools. The hardware can be described via a schematic diagram or by a language such as VHDL (VHSIC hardware description language, where VHSIC is an acronym of very high-speed integrated circuits). This technology has been in use for some years by manufacturers of large scale audio equipment such as Fairlight and Calrec [Kanzler, 2007; Warrington, 2010]. It is quite feasible for individuals to quickly devise and implement their own audio hardware processing designs. Whilst it might be fanciful to presume that this complete approach to the development of bespoke processors in hardware as well as software will be widely adopted, it does offer the possibility for the sound recordist to develop tools adapted to their particular recording environment and situations. Whilst this combination is probably quite uncommon in those who would currently refer to themselves as recording engineers, there are degree courses which offer the study of digital electronics alongside recording techniques (such as at the author’s institution). Configurable, digitally controlled analogue systems are also available in the form of Field Programmable Analogue Arrays (FPAAs), although these are typically employed where digital solutions are not cost-effective [Macbeth and Roberts, 2004]. Perhaps in this sense things have come full circle: digital technology now offers cheap and accessible analogue design and realisation for audio.

Summary
This paper has set out to present thoughts on where, and how, the recording 'engineer' figures in the digital recording studio and how the role has changed since all-analogue recording was the norm. If engineering is the application of scientific knowledge and numeric ability to solve problems, then the ubiquitous presence of the computer for audio recording and processing offers a wide range of possibilities for nuanced calculation of solutions to problems typically found in the studio. But, with such transparent recording devices and so many preset solutions to recording tasks, there is also the danger of relegation to the role of mere operator – someone who keeps the machinery running and follows recipes whilst the creativity, in all senses of the word, rests elsewhere both inside and outside of the studio. Perhaps there are many situations where this scenario is cost-effective and satisfactory but this need not be the case everywhere. This author has first-hand experience of situations in which digital technology has offered a means for engineering solutions to problems (quickly building a time-shift panning device, coming up with an alternative when a time-stretching algorithm was creating quite noticeable artifacts) which would have been prohibitive, in terms of time and cost, to achieve via analogue means. One may agree with Ray Dolby, that "all major problems were solved long ago" and wonder if there is a place for an engineer within recording, or one may acknowledge that the individual challenges of each recording scenario require new solutions to a lesser or greater extent. If it is the latter, then the sheer flexibility and generality of 'digital' within the studio offers the chance for engineering to express itself even more effectively than before.

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