How Does Engineering Bridge into the Traditionally 'Creative' Realm of Music?

Part of a Series on How Engineering Supports the Creative Industries and Arts

BY BRIDGET A. MOORMAN, *ARIZONA BETA '85*

Introduction

Writing, making, and recording music is considered one of the more creative endeavors or industries as compared to most traditional engineering industries. The technology used in recording music has evolved from trying to capture a 'live music performance' in order to hear that performance repeatedly, to capturing sounds or musicians playing as separate 'data items' and integrating them into a unique expression that is pleasing to the human ear. From an engineering standpoint, the basic functions of signal transduction, processing, integration, and archival involved in music production are the same. However, as with all industries, the advent of the computer and ubiquitous telecommunications have expanded the options available within each of those functional aspects. (See sidebar by Beato, page 9, on the twenty top inventions that changed music).

The aesthetic and subjective property of music being pleasing to hear places music production in the creative realm.

According to Geoff Foster, Grammy Award winning recording engineer and chief engineer at Air Studios in London, UK: "The production of a song is a form of storytelling and recording engineers help in that creative process. As a recording engineer, the task is to facilitate getting a great recording and that means knowing where problems lie in a particular process — it is not just about the gear. Moreover, the engineer needs to understand and prepare a situation where problems don't impede the creative process. This is done by creating an environment where the musician feels safe and willing to make that outpouring of their soul to the performance. The understanding of this human side of the recording process is a significant part of recording engineering; perhaps the most significant."[1]

In the sound recording industry, there is a difference in the aim of the recording system from most systems in that from a creative or aesthetic standpoint, the system is an 'artistic' tool allowing distortion of the sound signal at each component adding an auditory desirable and pleasing aspect to the final product. While many times a 'clean' signal may be desired at the aggregation point, beyond that, music can involve significant audio signal manipulation in the final product. The "three basic components of an existing audio signal that can be manipulated are: **frequency** (gain at given frequencies adjusted using an equalizer); **dynamics** (done through signal compression or expansion); and, **time delay** (done through the application of reverberation or the introduction of echo and the subsequent harmonics). A combination of all these three is done in sound processing," said Foster.

Briefly, frequency changes are self-evident, i.e. changing the spectral content of the sound that is captured and controlled through adjusting specific frequencies using an equalizer. Dynamics management refers to compression or expansion of the sound waveforms so that the loudest and softest signals either have less or more difference between them than when they were recorded, creating balance amongst the song's constituent sounds. It also allows certain sounds and sound characteristics to come to the forefront or the background as desired by the creator. Time delay and reverberation (reverb) are used to simulate location based aspects of sound with placement and repeating of echo characteristics and can be done via the electronics or the space used for recording. As an example, "when recording a live orchestra with many instruments (in a large space) versus one instrument (in a small space), the sound recorded is captured along with reflections and colorations to the frequency response caused by interaction with the environment. A sound source far away in a big space will have less high frequency information as that information is lost traveling through the air, yet will have more reflections recorded at a comparable amplitude," said Foster.

The figure on page 6 depicts the way in which a sound signal can travel through the different aspects of music/recording production. The musician is in a room playing an instrument emitting a sound wave which is detected or picked up by a sound transducer (microphone). That signal is then amplified and captured on tape and/or a Digital Audio Workstation (DAW) for further processing and integration. The final 'music product' is then archived onto a media which can be played at any time desired through speakers or headphones to recreate the produced and processed sound waves which is then heard by the human ear.

PHYSIOLOGY OF THE EAR HOW THE BRAIN PROCESSES SOUND

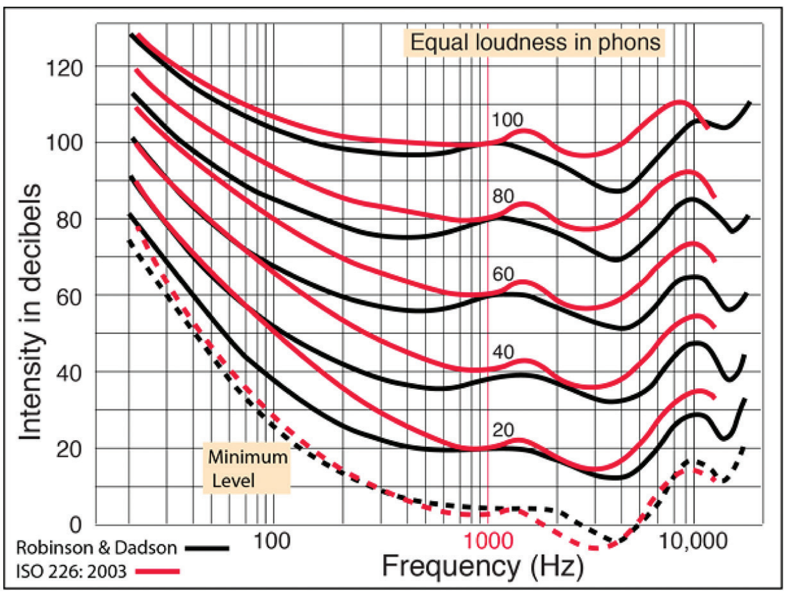
"A sound wave is an air pressure disturbance that results from vibration."[2] The human ear detects both the sound pressure level (SPL) and frequency of that sound wave. The audio frequency range for the human ear is 16Hz-20kHz with the best adaptation between 1kHz-3.5kHz (human speech is between the 80-260 Hz range). The sensitivity of hearing varies across this range as a result of the resonance properties of the ear and neural processing of the sound signal. A decibel represents the logarithmic ratio of the SPL as compared to the baseline or threshold detection of that sound pressure in the human ear. However, due to the frequency sensitivity variation (diminishment of detection) at the edges of the human hearing range, the perceived loudness of certain frequencies is a complex relationship defined by the loudness parameter phon.[3] A representation of a unit phon and the graph of the hearing range of the human ear as a function of sound frequency and SPL is shown in **Figure 2** (ISO).

The human neurological processing lag from the initiation to detection of sound ranges from 6-10 ms.[4] This 10 ms sensitivity is also seen in auditory sound localization which represents the perception of "space and depth," i.e. sound waves and their reaction to the environment. [5] "By modifying the space when recording the different sounds separately and then integrating them, the final product can give the impression of being performed 'live' in the same room at the same time (right back to the original intent of the design of recording systems)."[1] This is done by judicious use of reverb effects as well, mimicking the natural echoes one hears when listening to sounds live. Another example is use of a sound for tuning reference. If the brain's first information is a tuning reference, then a listener will use that reference for the tuning & harmonic relationships that follow. However, if that frame of reference changes, it changes the perception of what is in tune.[1] Lastly, the human processing of sound depends on a baseline that is initially established with regard to the Ambient sound. For example, when recording classical music before the music begins, there is often Ambient noise. If heard before any pitched information, the noise is perceived by the brain. If that noise is trimmed off and the first auditory information is pitched, the brain will not place as much emphasis on the noise and perceive it less. However, some noise is desirable as most humans find Digital Silence disturbing.

MICROPHONES

The microphone is as much an instrument as a musical instrument because a sound signal can be modified (distorted) at the microphone input.

Figure 2: ISO 226:2003 set of equal-loudness curves and the Robinson & Dadson curves with phon line depicting equal loudness perception at different decibels for 1000 Hz (phon = dB sound pressure level (SPL) at 1kHz).[11]
From: http://hyperphysics.phy-astr.gsu.edu/hbase/Sound/eqloud.html



Collection of microphones: (left to right) Ribbon, two dynamic, and a condenser.

Tube sound amplifier. Guitar amplifier, preamp, and power amp detail, gain and tone control knobs.

A microphone is a transducer that converts sound pressure waves to a voltage using a diaphragm that oscillates (similar to the eardrum). How that diaphragm is controlled, as well as the desired signal outcome, determines the type of microphone used to capture a sound. The three types of microphones predominately used in music recording are dynamic (or coil-based), condenser (or capacitive), and ribbon. [6]

A dynamic microphone has the diaphragm connected to a coil of wire wrapped around a magnet and as the coil moves due to the diaphragm movement and across the magnet, a voltage is created on the wire. Speakers work on a similar principle but in reverse (transducing varying voltages to pressure waves for sound). Dynamic microphones are simple, durable, and passive designs. They are high mass, i.e. can handle higher sound pressure levels (dB) without damaging the microphone or adding unwanted signal distortion. They are omnidirectional or cardioid polar (capture sound from the front and sides and not the rear of the microphone). They are predominately used in live music situations and with louder sounds (such as guitar amplifiers, drums, or loud vocals). They also tend to filter out ambient noise better than condenser or capacitive microphones.

However, dynamic microphones are not best for capturing high frequency or low loudness sources of sound, which require higher sensitivity, so a condenser or capacitive microphone is used. The electrically conductive transducing diaphragm in this microphone is aligned with a metal plate and as it oscillates the distance between the two metal plates change, thereby changing the capacitance between the plates and creating an electric signal. This signal needs 48V of 'phantom power' to power the audio electronics downstream, so it is not a passive detection system. These microphones are generally used in studios with good acoustic characteristics (not too much reflection) as they are sensitive and can detect subtle sounds as well as vocals and acoustic instruments. They have a fixed polar pattern of cardioid or omnidirectional and at times two diaphragms can be mounted closely together in the microphone allowing audio signal cancellation or mixing at the microphone input versus just downstream.

The third type of microphone is a dynamic 'ribbon or velocity' microphone which uses very thin corrugated metallic ribbon suspended in a magnetic field which moves with sound pressure and produces an electric signal. It is low mass leading to an excellent frequency response. It is bi-directional, so sound is picked up mainly on the flatter sides of the ribbon with minimal pickup of 'side' sounds. This could be useful in 'capturing a choir' allowing for the choir and reverberation in the room to be heard. Lastly, the SPL response is non-linear mimicking the human ear's response resulting in a more natural sound. [7]

AMPLIFIERS

The sound signal, which is now electrical, travels through to an amplifier. The amplifier enables signal energy intensity equalization on the track (digital or analog) due to the different qualities of the initial sound signals. This signal amplification can occur at the microphone and/or before the track recording. There are three types of amplification: pre, low-power, and power. Pre-amps are used for weaker sound signals increasing their power to levels that can be manipulated in the downstream processing equipment. Low-power amplifiers are primarily used to modify the sound signal in volume and frequency, often shaping the signal for aesthetic and mixing means. Power amplifiers are used to increase the sound signal intensity and drive speakers in the reproduction of the sound for hearing. All amplifiers endeavor to minimize noise while amplifying the signal.[8]

DIGITAL AUDIO WORKSTATION

All of the signals are then aggregated and managed in a Digital Audio Workstation (DAW), a software package which allows the manipulation of the recorded sound using digital sound tracks (analogous to the analog tape tracks used in the past). Each track represents a recording of a discrete sound signal. The DAW can aggregate, integrate, and process the different sound signals providing a variety of sounds for different engineers/music creators to develop a final layered music product. This flexibility, with regard to the ability to process and mix the different sounds captured, can add depth and individuality to the music produced. As above, it is here that the audio signal can be modified with regard to frequency, dynamics, and time delay. In the past, this was done by manipulating the sounds with specifically crafted electronic units designed to distort the sounds in different ways and then assembling the different musical elements along side each other by recording on analog multi-track tape recorders. Sometimes the tape media was run backward for that sound manipulation.

ARCHIVAL

The final product or song is then 'copied' onto a readable media which is then distributed or sold to be played by various systems for a listener's pleasure. Interestingly, with the preponderance of digital production today as well as use of many digital sound samples, data management has become an issue. Some sampled songs can require over 1TB of storage. Moreover, archival media has evolved (5-10 years maximum for many media) and as the songs get transferred to another media and/or degrade while in storage (or lost), there is a concern that a lot of sounds and songs have been or will be lost.

ANALOG/DIGITAL/ ANALOG-DIGITAL HYBRID

As with most industries, computers have allowed for more higher quality outputs by individuals efforts (disaggregation of the knowledge or functionality) than was possible before. However, the best quality still comes from use of a 'professional' studio for more data capture and layering of those sounds into a musical product. Also, as in all other industries, the advent and use of the computer brought about much change in the music production process and end-product. In 2003, Dr. Art DeLagrange, *MA B '62*, shared some prescient thoughts on the use of computers in the engineering of music for *The Bent*:

"Aside from radio and recording, electronics was first used with microphones to amplify the voices of solo singers, eliminating the need for operatic voices. Next, it was used for instruments whose sound was not loud enough, such as a guitar... (and) mounting the microphone directly on the guitar to reduce extraneous pickup. But because the strings were usually metal, the pickup need not be acoustic at all, but could be magnetic... (which is) insensitive to ambient sounds directly. (Now) the pickup is an integral part of the instrument, not an addition...the logical extension of the electronic organ was the music synthesizer, basically a computer connected to a keyboard (which) can reproduce the sound of most common instruments, including a chorus of human voices (singing "aaahh"), plus unique sounds of their own. Often you are not hearing the instrument you think you are, but a synthesizer... . (P)erfection is usually achieved electronically; when you hear "one" recording you are usually listening to a composite of a dozen or more "takes" ... . The frailty of human nature cannot compete with the computer. The promoter of a mega-concert cannot cancel because one singer has laryngitis; that part will be "lipsynced," if not the entire performance. If the sound all comes out of loudspeakers anyway, what's the difference?"[9]

Digital Audio Workstation: Mixing board in sound recording booth with computers.

Rick Beato: Top 20 Inventions that CHANGED Music

1) **Music notation/printing press**
2) **Radio**
3) **Internet**
4) **Television**
5) **Keyboard/Harpsichord/ Piano – Standardizing how musicians communicate**
6) **Compact Disc (digitization of music storage in portable form-media – 74 minutes of music)**
7) **LP/Vinyl Records**
8) **Personal Computer**
9) **Headphones**
10) **Walkman**
11) **Electric Guitar**
12) **IPOD**
13) **Social Media – disaggregation and decentralization of the control of the musical product**
14) **Phonograph – recording – disaggregation to time for the musical experience**
15) **MIDI standard – musical instrument digital interface; 16 channels - connect piano keyboard to computer (or any instrument)**
16) **Digital Audio Workstation (DAW) – most common application is PROTOOLS**
17) **Vacuum tube – tube amplifier**
18) **Magnetic tape – recording/ reproduction**
19) **Napster**
20) **Fuzz Petal**

Notice that most of these inventions are due to the invention of electronics and telecommunications and they brought about the ability to modify, store, and decentralize aspects of music making and distribution. They also brought about the disaggregation of time and location for the making, performing, and distribution of the musical product. From a business perspective, they also allowed the musician the ability to have more control over their intellectual property if they so desire. From a consumer point of view, the inventions provide music portability and the ability to disaggregate, collate, curate, and store their music collection.

"THE UNDERSTANDING OF THE HUMAN SIDE OF THE RECORDING PROCESS IS A SIGNIFICANT PART OF RECORDING ENGINEERING; PERHAPS THE MOST SIGNIFICANT" — GEOFF FOSTER

Some music produced today is deemed 'mediocre' in some sense because it has been digitized and quantized so much that distortion that might be pleasing to the ear is lost. When dealing with a creative and aesthetic endeavor, the human response is key and varies individually. However, it seems that the organic nature of that response can at times prefer the analog technology in creative expression. As part of an answer to this loss of aesthetic value (fidelity and warmth), there is a resurgence of the use of some analog components in the music production. For example, "the saturation curve of analog tape is non-linear (as the sound placed onto/off the tape), which distorts the sound (compresses it) in a subtle way that adds a warmth and breadth to the human perception of that sound."[1] To attempt to replicate this warmth and range with digital systems, the sample rate is increased to create sounds that are closer to an analog system; however, there is still loss in the lower end of the range; moreover, many times with over-processing, one can hear the digitization. Many recording engineers and musicians prefer to use vacuum tube amplifiers with some of their musical instruments when recording stating they prefer the 'warmth' an amplifier brings to their sound creation. Moreover, there is an increased desire by many in the music industry to release recordings on vinyl disks again, in some cases with vinyl sales exceeding digital media. With computers, one can end up trying to optimize too many variables and the end-product isn't satisfying. Constraints can help focus the creative process, thus the desire for some creators to have hybrid analog-digital components in their recording systems.

SUMMARY

What engineers design is usually a crude approximation of designs seen in nature, hence, the constant evolution of technology to try reach the pinnacle of natural designs. From the beginning of recording systems, we have moved from an attempt to 'capture' a live performance of a full cohort of musicians and instruments to capturing single musical efforts or mini sounds which are aggregated/integrated into a new sound product, which sometimes try to replicate a live performance. Interestingly, as music production has become more technical, the desire to have a more organic and ear-pleasing sound has re-emerged, leading to a hybrid digital-analog environment for sound production. In the end, an engineer who uses the principles of the physical world to build tools that are useful to humans also builds the tools that enable humans to creatively express themselves.

BRIDGET A. MOORMAN, BSME, MSBME, COL, USAF (Ret.) has 30+ years' experience in the clinical engineering/digital health field working and consulting for large healthcare organizations as well as serving as chair of the board of examiners for certification in clinical engineering (2011-13). She has worked in bio-mechanical research, power-line relay design and space system launch, and telecommunications. As an Air Force reservist, Bridget served as IMA to the MILSATCOM Directorate. Her B.S. from Arizona State University (ASU) is in ME; M.S. in BME from Hartford Graduate Center/Rensselaer Tech; and MAS in healthcare informatics ASU College of Health. Her professional awards include the 2019 ACCE/HIMSS Excellence in Clinical Engineering and Information Technology Synergies Award. She is an advisor to the Tucson Alumni chapter (2015 -) and an Instrument Private Pilot, Single Engine Land.

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