Assignment Title: Analogue and Digital Audio Report

Module: VEPT10018 Sound and Audio Theory

Name: Nathan McCubbin

N0708166

Contents

Introduction

This report will assess the science behind analogue to digital conversion of audio by looking in to four different aspects of it. Acoustical to Electrical Conversion, Analogue Signal Principles, Analogue to digital signal conversion and Digital Principles. By going through books, articles and experiments to find the research.

1. Analogue, Electrical, Conversion and Digital.

1.1 Acoustical to Electrical Conversion

To truly understand the conversion of sound, it is good to know what sound is. Sound is produced by the vibration of the air molecules, by compression and rarefaction of the molecules which in turns develops a soundwave. (Figure 1.1) (Howard, 2017) (Pohlmann, 2010) (Rumsey, 2012). An example of this is when a snare drum is struck. As when the snare is hit, the air surrounding it will become disturbed as the skin of the snare is moving back and forth (vibrating) which causes compression, from the skin moving forward, and rarefaction with the skin moving backwards. (Howard, 2017) (Pohlmann, 2010) (Rumsey, 2012).

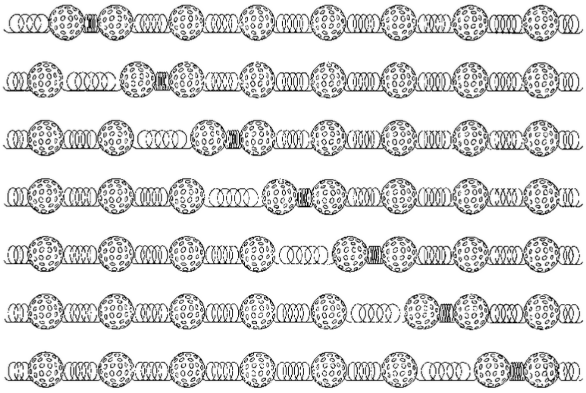

Figure 1.1 Compression and Rarefaction (Howard, 2017)

So, the devices that are used to convert the acoustical energy is transducers. Transducers receive the energy from a source, examples of this is a drum stick, when struck on the snare the mechanical energy taken from the stick will become acoustical energy. There are many kinds of transducers but the one of importance is the transducers that take acoustical energy and transform it in to electrical energy. (Pohlmann, 2010)

A transducer that is used to convert acoustical energy to electrical is the microphone, as when the sound wave meets the microphone, the diaphragm within the microphone which in turn creates an electrical current that flows through the microphone. Which is the conversion of the acoustical energy. (Huber, 2017) (Figure 1.2)

Transducer	From	To
Ear	Sound waves in air	Nerve impulses in the brain
Microphone	Sound waves in air	Electrical signals in wires
Analog record head	Electrical signals in wires	Magnetic flux on tape
Analog playback head	Magnetic flux on tape	Electrical signals in wires
Phonograph cartridge	Grooves cut in disk surface	Electrical signals in wires
Speaker	Electrical signals in wires	Sound waves in air

Figure 1.2 Examples of transducers (Huber, 2017)

1.2 Analogue Signal Principles

After compressing the molecules, it will then spread out further to the next low-pressured molecules and continuing until all the energy is used up, then reverting to its original form. (Huber, 2017)

Compression of the air is when the vibrating mass is activated, by striking a guitar string or singing a vocal line, it will scatter from the original resting position, choking the air molecules within it's radius into a compact area, distance from the sound source and causes there to be a dense atmospheric pressure. (Huber, 2017) (Howard, 2017) (Figure 1.4.A)

While the opposite of compression is rarefaction. Which is where vibrating mass moves inwards from the original position, which is the position created of lower atmospheric pressure rather than the normal state. (Huber, 2017) (Howard, 2017) (Figure 1.4.B)

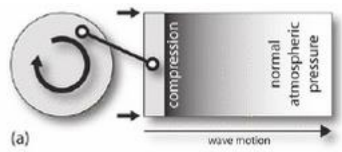

Figure 1.4.A Compression (Huber, 2017)

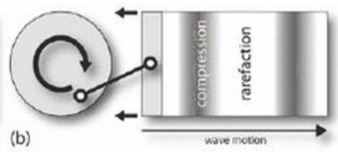

Figure 1.4.B Rarefaction

As Amplitude is associated with acoustical sound, voltage is the equivalent of amplitude in the terms of electrical sound, (Figure 2.3) shows that if the voltage and acoustical energy were to be both measured that they would both turn out the same. As the acoustical compression is the same as the positive voltage of the electrical form and it would be the same for the rarefaction and negative voltage, both would be the same. (Rumsey, 2014)

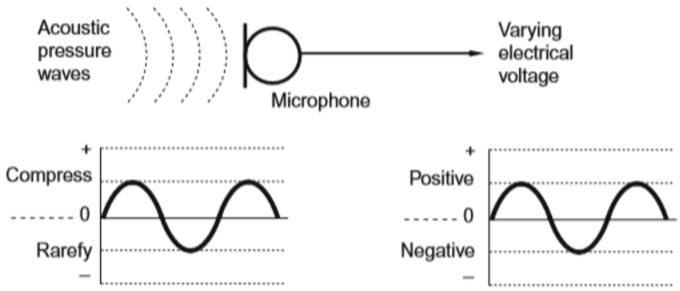

Figure 2.3 Compression & rarefaction equivalent to positive & negative (Rumsey, 2014)

The other part of the electrical sound that is important is the current, as current is the electrical equivalent of the air particles motion. As in acoustical terms, the sound wave traverses through the air molecules, that is the same as the electrical sound wave traverses through the motion of the electrons which are integrated within the metal of the wire (Rumsey, 2014).

As the voltages turns in to a positive one, the current goes down a direction while once it becomes a negative voltage, the current changes to another direction. The voltage that is produced by the microphone is constantly switching from positive to negative and back again in sync with that of the compression and rarefaction of the acoustical sound wave. The current changes it's direction every half-cycle like the original sound wave. Since the electrons carrying the current don't go to a place, they end up oscillating around a fixed point, which is known as alternating current. (Rumsey, 2014)

1.3 Analogue to Digital Signal Conversion

Converting an analogue signal to a digital one is done by measuring the amplitudes of the analogue signal at certain points within the time of it's cycle, also to assign a binary digital value to each measurement. These steps are known as sampling and quantizing, in that order. (Rumsey, 2014)

One of the tasks integrated within the design of a A/D convertor (Analogue to Digital) is to convert an analogue into binary codes. Sampling is done by measuring of the amplitude of the audio waveform at certain points in time (Figure 1.5) (Rumsey, 2014) (Huber, 2017)

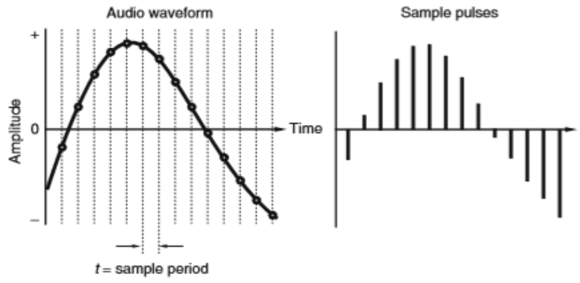

Figure 1.5 Sampling of an analogue signal. (Rumsey, 2017. PG: 241)

An interesting way to explain such a concept like sampling is by understanding it like movies. What you see on the screen is frames. Still images taken from a camera, but thousands of these pictures places together in order of the scene in place to produce a moving image. (Huber, 2017) (Rumsey 2014)

Quantizing is a process of digital conversion that takes place after sampling is done. Which is where the sampled audio signal's sampled amplitudes points are mapped on a scale of stepped binary values. (Figure 1.6)
Which in turn means that the quantizer finds the mid points of the sampled amplitude that don't lie on a fixed number point and will move it up and lower the section of audio to a fixed number. This process is done so that each sampled amplitude can be shown by a unique binary number in pulse code modulation. (Rumsey, 2014)

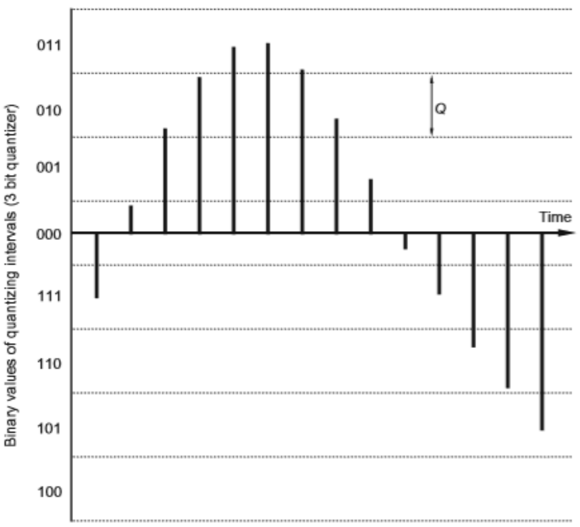

Figure 1.6 Quantization of a Sampled Audio Signal. (Rumsey 2014. PG: 250)

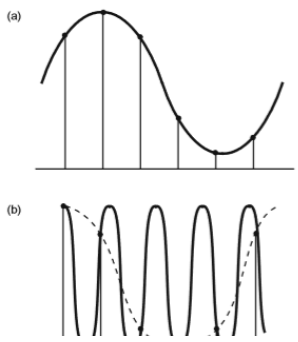

Figure 1.7 Result of too few samples taken. (Rumsey 2014. PG: 245.

If they are too few of samples taken from an analogue audio signal, then there will almost be gaps within the sampled signal. (Figure 1.7) which will result in the converter taking the gaps and filling it up with something new. Examples of this is Figure 1.7, as it has taken a sine wave, that could be registered at a mid-frequency band and converted it to a higher-frequency sine wave. (Rumsey, 2014)

As stated early within this section, a A/D converter turns an analogue audio signal in to a digital one, which is a sequence of binary numbers. So, in a binary system each bit shows a power of two. (Figure 1.8)

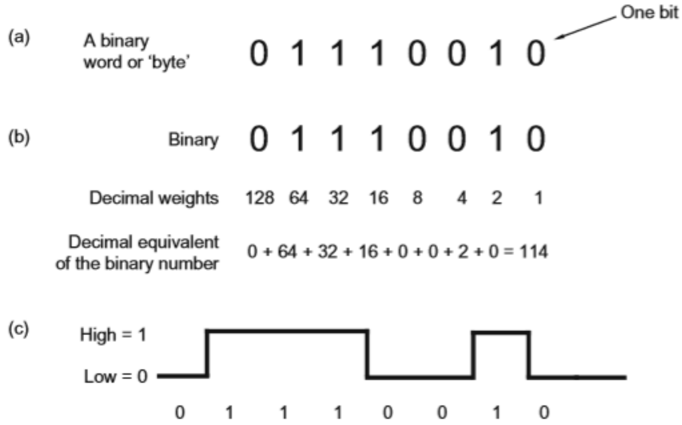

Figure 1.8 Binary (Rumsey, 2014. PG: 235)

A number that consists of more than 1 bit is known as a binary word. Which there are different types of these words. Like Bytes, which is an eight-bit binary word and a nibble, which is four-bit binary word. The more words there are, then there is a larger amount of numbers of states that are represented. (Rumsey, 2014)

As shown in Figure 1.8, the way computers understand the vast amount of binary presented is by adding an overall values of the set of binary code, as each bit within the binary code has a value, that goes up in doubles, (1 + 1 = 2, 64 + 64 = 128) and when the binary code has some set ones within the code, that is when the value is used and will be added the overall value. So, within the Figure 1.8, the binary code has four 1s present, which gives them value depending on their placement. Which adds to: 2, 16, 32 and 64. The system would then add up these numbers to find the overall value of 114. Which after the overall value is found, the system will know the value of the binary code. (Rumsey, 2014)

Other examples of finding the overall value of a binary code. (Appendix, 1) 1111, individual values: 8, 4, 2, 1. Which together equals 15. 1001, individual values: 8, 0, 0, 1. Which together equals 9. The reason why they are different is because 0 has no value. (Rumsey, 2014).

Dithering is the effect of cancelling quantizing distortion using a noise signal through-out the signal. Thus, allowing the converted audio to fade out without hearing a distorted sound left in its place. These noise signal dithering is considered more desirable rather than a low-level distortion. Even with the case modern high-specification convertors noise floors are typically inaudible. (Figure 2.0) (Rumsey, 2014)

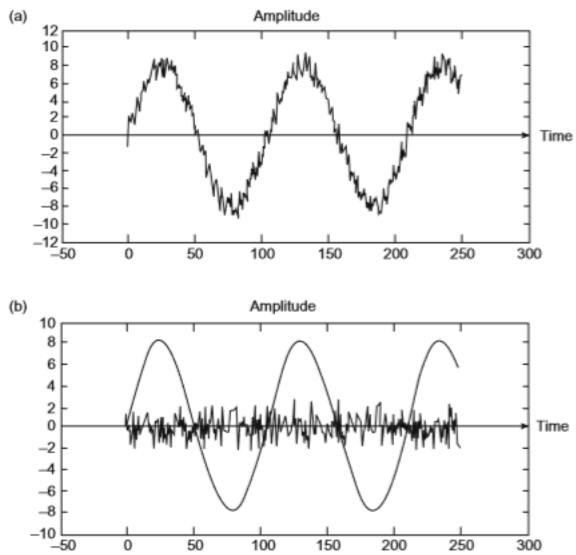

Figure 2.0 The effect of dithering. (Rumsey, 2014. PG: 259.

It is known that distortion from an undithered A/D converted audio signal is the result in a cooperation of the signal its self as well as quantizing error. Resulting in an unwanted outcome. Thus, adding dithering, (Noise) which is generated at a unpredictable signal, has the deliverance of having the unpredictability placed on the quantizing error as well, which converts it to noise along side the dithering. This random signal effect is averaged out, resulting in a noise quantizing error and fixed noise floor. (Rumsey, 2014)

1.4 Digital Principles

With analogue recordings, as good as it can be, there are common unwanted qualities that can persist. For example, the fine of unwanted and wanted signals is difficult for the playback system to differentiate a difference. And these unwanted signals can be such elements as distortion, noise and any other forms of interference that appeared during the recording process. (Rumsey, 2014)

While arguments over analogue and digital signals may never cease. It is digital signals that are higher in quality. Because digital converts an electrical signal produced by the microphone and transforms the electrical signal in to series of binary numbers, which each of the binary numbers represent the amplitudes of the signal during anytime. The signal being generated as a binary system, allows the system to notice whether the playback signal is correct. (Rumsey, 2014)

One of the fundamentals of Digital Audio is the law of the Nyquist Theorem, which states that; with the desire to have the frequency bandwidth accurately be converted in to the digital realm. The sample rate needs to be at bare minimum double the amount of the highest frequency to be recorded. (Huber, 2017)

Frequencies that are higher than the sample rate will have an issue of being accurately captured, resulting in the sample not being able to capture the higher frequencies which would record alias frequencies that do not exist. Which causes unwanted harmonic distortion. (Figure 2.1) This would practically mean if the frequency bandwidth is 20kHz then the sample rate needs to be at least 40kHz. (Huber, 2017) (Pohlmann, 2010)

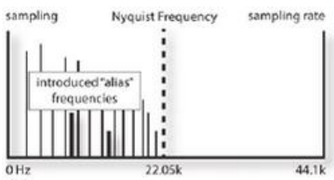

Figure 2.1 Alias (Huber, 2017: PG 200

To combat the effects of aliasing, a low pass filter is applied to the circuity because the process of sampling is applied, and the result of this causes all the frequencies above the desired amount to not be recorded, giving a clear recording without any unwanted harmonic distortion.

But, if in theory the set sampling rate is 40kHz for the capture of everything below 20kHz, this would be result in the issue of desired frequencies being cut from the process. (Figure 2.2) Also, a hard cut of frequencies isn't immersive contract to the real workings of human hearing. So, the sample rate is raised to 44.1kHz, so a sloped low pass filter is applied to point of 22.05kHz. To solve the issue of losing frequencies. (Huber, 2017) (Pohlmann, 2010)

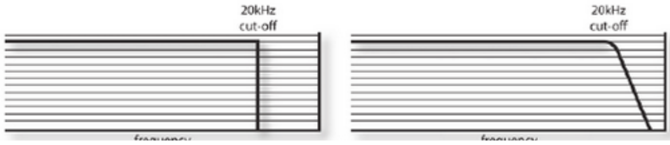

Figure 2.2 Low pass filter applied. (Huber, 2017: PG 201

The signal-to-error ratio process is done for the use of measuring the quantizing process. This process is similar to the analogue technique of signal-to-noise ratio but being a different concept as S/N ratio is used for the indication of dynamic range of an analogue system. The signal-to-error ratio within a digital system is used for the indication of the accuracy used to capture the sampled levels. (Huber, 2017) (Pohlmann, 2010)

The process of quantizing a signal into a digital word are not continuous, unlike analogue signals. So, since the number of steps that are encoded in a digital word can limit the accuracy of the quantization process, the digital word can only be a resemblance of the original analogue signal level. (Huber, 2017)

Conclusion

Within this report the goal was to simple, the access the science behind what makes analogue signals, digital signals, conversions of analogue to electrical, and analogue to digital. And this report has done just that, this report gives easily digestible information that isn't too hard to grasp.

A trend found between all four of these areas of signal conversion and principles of a signal is the fact of a middle man, it's simply not possible without it. For example, Acoustical to Electrical needs transducers like guitar pickups and microphones. Acoustical Signal Principles needs air molecules for the sound to travel. Acoustical to Digital needs, again, transducers like Microphone Preamps and Microphones. Lastly Digital Principles need the middle man of Nyquist Theory, as without this theory it would impractical to even convert the acoustical or electrical signal to digital.

Within this day and age, the most important section of this report to take away is Digital Principles, followed closely by Acoustic to Digital Conversion. Simply because digital is the next step of recording and producing music, subjectively speaking. So, understanding how digital works and how the transducers used to turn a signal digital is a key factor if the goal is to build a studio, a digital one at that. As knowing this information will navigate to better recordings and production. Needless to say that acoustical signal principles, and acoustical to electrical conversion is important too, in terms if the goal of the studio is to record instruments and vocals then it will best to understand these sections as it will help to build a better live space for your artist to perform within acoustically.

What is most significant is the what makes these four sections work and be practically used, as without these fundamentals working together, then a signal or conversion would be important to achieve. An example being: Sampling, if there was no sampling then the electrical conversion of an acoustical signal would either be impossible or terrible, as the transducer would instead make up a new signal which would be vastly different compared to the desired signal.

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3. Everest, F.A & Pohlmann, K.C. (2015). Master Handbook of Acoustics, Sixth Edition. McGraw Hill Books.
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Appendix

1. Binary Task: 1111 = 15, 1001 = 9, 1010 = 10, 11111111 = 255, 01011011 = 91.