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COMPARISON OF ANALOG-BASED PROCESSORS AND SOFTWARE-BASED PLUG-IN PROCESSORS

Analog Tabanlı İşlemciler ile Yazılım Tabanlı Plug-in işlemcilerinin Karşılaştırılması

Mehmet ÖZKELEŞ *

ABSTRACT

Today, DAW (Digital Audio Workstation) software lies at the heart of music production and has become an indispensable part of the industry. With the advancement of technology, music production is no longer confined to large studios but has also become accessible to individual users. Now, recording, mixing, and mastering processes can be carried out in a home environment using computer-based systems, without the need for expensive equipment. This fully digital workflow is referred to as "in-the-box" production. Technological developments have enabled individuals to produce music at a professional level in home studios, leading to the emergence of digital alternatives to analog devices. Digital processors have begun to replace analog equipment due to their cost-effectiveness and practicality. This study examines the differences between digital signal processors and analog devices in terms of usage, application areas, and sound processing methods. The research aims to compare digital and analog signal processors, particularly focusing on their processing methods, speed, flexibility, quality, and impact on sound. Based on the findings, it is concluded that digital signal processors, which are increasingly replacing analog processors in the music industry, do not possess the same characteristics as their analog counterparts.

Anahtar Kelimeler: Plug-in, Analog Digital Signal Processor (DSP), Music Technologies, Sound Recording Technologies.

ÖZ

Günümüzde DAW (digital audio workstation) yazılımları, müzik prodüksiyonlarının merkezinde yer almakta ve sektörün vazgeçilmez bir parçası haline gelmektedir. Teknolojinin gelişmesiyle birlikte müzik prodüksiyonu yalnızca büyük stüdyolarda değil, bireysel kullanıcıların da erişimine açılmıştır. Artık yüksek maliyetli ekipmanlara ihtiyaç duyulmadan, bilgisayar tabanlı sistemler kullanılarak ev ortamında kayıt, miks ve mastering işlemleri yapılabilmektedir. Bu sürecin tamamen dijital ortamda gerçekleşmesine "in the box" prodüksiyon adı verilir. Teknolojik ilerlemeler, bireylerin ev stüdyolarında profesyonel kalitede müzik üretmesine olanak tanımış ve analog cihazların dijital alternatifleri ortaya çıkmıştır. Dijital işlemciler, daha uygun maliyetli ve kullanışlı oldukları için analog ekipmanların yerini almaya başlamıştır. Dijital sinyal işleyicilerin kullanım şekli, uygulama alanları ve ses işleme süreçleri bakımından analog cihazlardan farkları incelenmektedir. Araştırma, özellikle dijital ve analog sinyal işlemciler arasındaki işleme yöntemi, hız, esneklik, kalite ve ses üzerindeki etkilerini karşılaştırmayı amaçlamaktadır. Bu bulgular doğrultusunda, müzik endüstrisinde kullanılan analog işlemcilerinin yerini almakta olan dijital tabanlı sinyal işlemcilerin, analog cihazlara kıyasla aynı niteliklerde olmadığı sonucuna varılmıştır.

Keywords: Plug-in Analog İşlemciler, Dijital Sinyal İşleyiciler (DSP), Ses Kayıt Teknolojileri.

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GENİŞLETİLMİŞ ÖZET

Bu araştırmanın temel amacı, ses kayıt sürecinde kullanılan analog tabanlı sinyal işlemciler ile yazılım tabanlı (plug-in) sinyal işlemcilerden elde edilen verilerin teknik ve estetik açılardan karşılaştırmalı analizini yapmaktır. Çalışma kapsamında, özellikle gitar ve vokal sinyalleri üzerinden elde edilen çıkış seviyeleri, frekans davranışları, dinamik tepkiler ve satürasyon farklılıkları gibi çok boyutlu kriterler incelenerek, her iki sistemin avantajları ve sınırlamaları belirlenmiştir. Araştırma, nitel araştırma yöntemlerinden biri olan doküman incelemesi yöntemi kullanılarak gerçekleştirilmiştir. Analog tabanlı sinyal işlemciler ile yazılım tabanlı işlemcilerden elde edilen ses verileri hem görsel analiz (VU metre, spektrum ve spektrogram çıktıları) hem de işitsel değerlendirme yöntemiyle karşılaştırılmıştır. Bu araştırmada inceleme, gitar ve vokal sinyalleri üzerinden yürütülmüş; özellikle miksaj sürecindeki ses karakteristikleri, harmonik yapı, frekans dağılımları ve çıkış seviyeleri dikkate alınmıştır.

Bu araştırmada elde edilen bulgular, analog tabanlı ve yazılım tabanlı sinyal işlemcilerin ses mühendisliği süreçlerine etkisini çok yönlü biçimde ortaya koymaktadır. Analog sistemlerin sunduğu harmonik zenginlik, özellikle vokal ve gitar gibi karakteristik enstrümanlarda, müzikal dokunun daha sıcak ve doğal algılanmasını sağlamaktadır. Bu yönüyle, analog işlemciler; caz, blues, rock gibi dinleyicinin organik ses beklentisinin yüksek olduğu müzik türlerinde tercih nedeni olmaktadır. Buna karşın, yazılım tabanlı işlemciler; elektronik, pop ve dijital odaklı prodüksiyonlarda sundukları netlik ve kontrol kabiliyetiyle ön plana çıkmaktadır. Dolayısıyla tercih kriterleri yalnızca teknik özelliklere değil, aynı zamanda estetik hedeflere ve tür bazlı stil anlayışına da bağlıdır.

Analog donanımın yıllar içinde edinilen ses karakteri, çoğu zaman nostaljik veya kültürel bir beklentinin de karşılığı hâline gelmiştir. Bu bağlamda, vintage tüp kompresörler veya FET tabanlı klasik donanımlar, belirli bir ton beklentisini karşılamak için hâlâ stüdyo standartları içinde yer almaktadır. Ancak günümüz yazılım teknolojileri, bu donanımların dijital emülasyonlarını geliştirerek daha erişilebilir çözümler sunmayı hedeflemektedir. Buna rağmen, yapılan spektrum analizlerinde bu yazılımların hâlâ birebir benzer harmonik yapı ve frekans yoğunluğu üretemediği görülmektedir. Bu farkın en belirgin nedenlerinden biri, analog donanımın sıcaklık, elektrik dalgalanması ve devre karakteristiği gibi fiziksel etkenlerden doğan benzersiz varyasyonlarıdır. Yazılım tabanlı sistemler ise genellikle sabit algoritmalarla çalışmakta ve bu değişkenliği doğal olarak yansıtamamaktadır.

Buna ek olarak, kullanıcı deneyimleri de tartışmaya dâhil edildiğinde, birçok profesyonelin analog donanımın "dokunulabilir" fiziksel kontrol hissinden beslendiği, bu sayede daha yaratıcı ve spontane kararlar alabildiği belirtilmektedir. Diğer yandan, yazılım tabanlı sistemlerin otomasyon kabiliyeti, geri alma (undo) özelliği ve düşük maliyetli çoğaltılabilirliği, dijital çağın hızına uyum sağlamak adına vazgeçilmez araçlar arasında yer almaktadır. Bu da göstermektedir ki, her iki sistemin kullanımı sadece teknik değil, aynı zamanda ergonomik ve üretkenlik temelli bir seçimdir. Araştırma sonuçları, analog tabanlı sinyal işlemcilerin ses kayıt ve miksaj süreçlerinde daha doğal, sıcak ve dinamik açıdan zengin sonuçlar verdiğini ortaya koymuştur. Yazılım tabanlı işlemciler, pratiklik, maliyet avantajı ve esneklik sunmalarına rağmen, özellikle harmonik yapı, frekans yoğunluğu ve satürasyon davranışı açısından analog sistemlerin doğallığını birebir yansıtamamaktadır. Her iki sistemin sunduğu avantaj ve sınırlamalar, uygulama alanına göre değerlendirilmelidir.

Analog sistemler daha tutarlı ve organik sonuçlar verirken; yazılım tabanlı işlemciler ise dijital çağın sunduğu kolay erişim, otomasyon ve çoklu işlem kabiliyeti ile öne çıkmaktadır. Ancak, yazılım sistemlerin çoğu, analog sinyallere özgü ısıya bağlı değişkenliği, spektral esnekliği ve frekans enerjisini tam anlamıyla sağlayamamaktadır.

Bu çalışma sonucunda elde edilen veriler ve bulgular doğrultusunda, ses mühendislerine, yapımcılara ve aranjörlere yönelik bazı öneriler geliştirilebilir. Öncelikle, ses işleme sürecinde kullanılacak donanım veya yazılımın seçiminde, sadece ses kalitesi değil, aynı zamanda prodüksiyonun hedefi, bütçesi, müzik türü ve iş akışı dikkate alınmalıdır. Özellikle kısa teslim süreli projelerde, taşınabilirlik ve erişilebilirlik açısından yazılım tabanlı işlemciler tercih edilebilir. Ancak, yüksek kaliteli müzikalite ve harmonik doygunluk hedeflendiğinde, analog sistemlerin tercih edilmesi daha doğru sonuçlar doğurabilir.

Hibrit sistem kullanımının artırılması da önemli bir çözüm yolu olabilir. Örneğin, temel ton karakterini analog donanım ile elde ettikten sonra, dijital ortamda ince ayar ve efekt işlemleri yazılım tabanlı plug-in'ler aracılığıyla yapılabilir. Bu yaklaşım, hem analog tabanlı sinyal işlemcilerin sunduğu sıcaklık hem de yazılım tabanlı sinyal işlemcilerin getirdiği esnekliği birleştirerek daha etkili sonuçlar verebilir.

Ayrıca, yazılım geliştiricilerin, analog donanımın fiziksel ve termal davranışlarını daha detaylı simüle eden algoritmalar üzerine çalışmalarını yoğunlaştırmaları önerilmektedir. Günümüzde bazı ileri düzey modelleme teknikleri bu yönde gelişme göstermekteyse de hâlâ frekans yanıtları ve satürasyon davranışlarında gözle görülür farklılıklar mevcuttur. Geliştirilecek yeni nesil yazılımlar, analog karakteristiği daha doğru biçimde taklit edebilir ve dijital sistemlerin gerçekçiliğini artırabilir.

Akademik açıdan da bu konunun daha fazla deneysel veriyle desteklenmesi, farklı müzik türlerinde ve çeşitli donanım konfigürasyonlarında çok yönlü testlerle değerlendirilmesi önerilmektedir. Özellikle dinleyici algısına dayalı çift kör testlerin yapılması, işitsel farkındalık üzerine önemli sonuçlar sağlayabilir.

Son olarak, eğitim kurumlarında ses mühendisliği müfredatlarına hem analog hem dijital işlemcilerle ilgili uygulamalı içerikler eklenmesi, gelecek kuşakların bu sistemlerin avantaj ve sınırlamalarını deneyimleyerek öğrenmesini sağlayacaktır. Böylece öğrenciler yalnızca teknik bilgi değil, aynı zamanda tonal estetik üzerine de bilinç kazanmış olacaklardır.

The history of sound recording technology dates back to the 19th century and has evolved over time with technological advancements. In 1807, American physicist Thomas Young designed a device capable of recording acoustic signals onto a cylinder. In 1857, Léon Scott de Martinville developed a device called the phonautograph, which enabled sound to be recorded for the first time. Scott used an early method of capturing sound waves by darkening paper with the heat of a gas lamp, and on April 9, 1860, he successfully recorded a French folk song (Cumhuriyet Bilim Teknik, 1994). These developments marked a turning point in the field of sound recording and inspired many inventions in the following years.

In 1877, Alexander Graham Bell introduced his own sound recording device, further advancing this technology and laying the groundwork for sound engineering. That same year, Thomas Edison invented the phonograph, which he referred to as the "talking machine." The phonograph presented the first visual representation of sound by making it possible to process recorded audio (CNN TÜRK, 2008). Following Edison's success, in 1888, Alexander Graham Bell and Emile Berliner obtained a patent for the phonograph (Mc Queary, 1990).

The first attempts at magnetic tape recording were published in 1888 by Oberlin Smith, and in 1889, Danish physicist Valdemar Poulsen produced the first example of magnetic recording with a device called the "telegraphone" (Mumma, Rye, and Kernfeld, 2010).

At the time, microphones used in telephones contained carbon, which caused imbalances in sound and signal frequencies, leading to high background noise during music production, radio broadcasts, and recordings. As a result, telephone microphones were generally not used in music studios or broadcasting. Engineers at Western Electric developed a dual-button carbon microphone that significantly reduced these problems, making microphones usable in radio and music production (Burgess, 2014).

Mechanical recordings of the time caused acoustic power loss while managing sound frequencies, which negatively affected the quality by increasing background noise. In the early 20th century, engineers and scientists at Bell Labs began researching electronic sounds to advance recording technologies. Western Electric made a major leap by designing an electromechanical recording device using a condenser microphone. In 1925, Victor Talking Machine Company became the first company to conduct electronic recordings (Schmidt-Horning, 2013).

In 1955, Les Paul collaborated with Ampex to develop an 8-track recorder. In 1958, Ampex produced this device, and Les Paul's innovations paved the way for modern recording techniques. Therefore, 1958 is considered the year when multitrack recording devices were first manufactured (Önen, 2010).

Compressors first appeared in the 1930s in the U.S. with the Western Electric 110A series and rapidly spread worldwide (Pasinlioğlu, 2019). A compressor is a device that automatically controls audio (Bartlett, 2005). By compressing audio signals at a specific rate, it reduces the dynamic range between soft and loud sounds and balances output levels (Durmaz, 2009). Compressors that react to light are called optical (opto) compressors; the heat from the audio signal causes a light to increase in intensity, and when the signal level drops, the light dims (Coşaner, 2008). These types of compressors are widely used in sound engineering and recording because they enhance audio quality and offer a more consistent listening experience. The dynamic range control offered by optical compressors also plays a crucial role in live performances.

A limiter is a device that prevents the audio signal from exceeding a certain level (Edstrom, 2011). It prevents unwanted distortion and signal clipping during recording (Öcek, 2010).

An expander increases the lower-level signals, expanding the dynamic range and making quieter sounds more prominent. It emphasizes sounds below a certain threshold, increasing the overall dynamic contrast (Izhaki, 2008).

Today's audio systems commonly use sampling rates of 44,100 Hz and above. This is due to the upper limit of human hearing being around 20,000 Hz (Rumsey, 1994).

The music production process is not limited to composition or performance; it also involves the processing of sounds (Lefford, 2000). Over time, technological innovations have led to a major transformation in the permanence and quality of audio recordings.

With the rise of digital technology, discussions around analog versus digital methods intensified, and digital technology has become dominant in the music production landscape (Huber, 2010). Comparisons between analog and digital processors continue today, shaped by factors such as ease of use and cost. These debates influence the preferences of musicians and producers and play an important role in the evolution of the music industry. While the innovations offered by digital technology accelerate creative processes, some artists still prefer traditional analog methods. Digital signal processors (DSPs) began to be widely used in studios after the 1980s, offering more flexible and portable solutions than analog equipment. These digital systems, which process signals as numerical data, offer higher precision and accuracy and have become particularly advantageous in mixing and mastering for their error-free repeatability (Collins, 2009).

Starting in the 1990s, the development of computer-based music production software (DAWs – Digital Audio Workstations) made the music production process more accessible. Software such as Pro Tools, Logic Pro, Cubase, and Ableton Live allowed both amateur and professional users to carry out complex mixing and mastering tasks in their home studios (Senior, 2011). These programs were able to simulate the sonic character of classic hardware using digital plug-ins based on analog models.

Moreover, with the advancement of spatial audio technologies, sound production is no longer limited to stereo channels, making it possible to create multi-dimensional listening experiences. Technologies like Dolby Atmos, Sony 360 Reality Audio, and Apple Spatial Audio emphasize the spatial dimension of music, allowing listeners to feel as though they are "inside" the sound. With the proliferation of these technologies, sound engineers and producers now consider not only horizontal but also vertical sound spaces during the mixing stage (Rumsey, 2017).

This study examines the technical and technological transformation in music production that has occurred as digital signal processors (DSPs), which became widespread from the 1980s onward, have increasingly replaced analog processors.

Problem Statement

The advancement of technology has sparked a significant revolution in album production within home studios. A pivotal role in this development has been played by the digitization of analog devices and the evolution of digital signal processors (DSPs). Digital processors have become increasingly preferred due to their cost-effectiveness compared to analog equipment. This shift has led to a matter of preference between analog and software-based equipment in the music production process, driven by the influence of digitalization. This study examines the technical and technological differences that arise from the use of software-based signal processors in place of analog-based ones during the recording, mixing, and mastering stages. These differences can have considerable impacts on factors such as sound quality, processing speed, and cost. Furthermore, innovations brought by digitalization have enhanced musical creativity, allowing producers to create sound designs that were previously unattainable.

Research Question

What are the differences between analog-based processors and software-based plug-ins in the sound recording process?

Purpose of the Study

The purpose of this study is to compare the data obtained from analog-based processors and software-based processors used in the sound recording process.

Significance of the Study

A comparative analysis of analog-based and software-based (digital) processors based on multidimensional technical and aesthetic criteria—such as sound quality, processing performance, control and flexibility capacity, cost-effectiveness, compatibility with technological advancements, and sonic characteristics—is of great importance in determining their suitability for different usage scenarios. Such a comprehensive evaluation will serve as a guiding resource for professionals working in the field of sound technologies, musicians, producers, and academic researchers, helping them consciously choose the systems that best meet their needs. Additionally, this resource may form the basis for raising industry standards and developing innovative solutions. By achieving optimal performance both technically and aesthetically, sound engineers and arrangers can achieve more effective results in their projects.

Limitations of the Study

This study is limited to digital audio workstations and their components.

Assumptions

It is assumed that the signal processors used during the production process in this study reflect their actual performance characteristics based on the measurement and interpretation of quantitative data obtained through spectrum analysis.

METHOD

Research Design

This study, which aims to reveal the similarities and differences between the software used in today's computer technologies and analog devices, is a descriptive research based on the general survey model.

Survey models are research methods that aim to describe a situation that existed in the past or still exists in the present, just as it is. The subject of the research—be it an event, individual, or object—is described within its own conditions without any alteration or manipulation; there is no intent to change or influence it. Although survey models can be used independently as a research method, a research model that solely relies on a survey is not considered sufficient by itself (Karasar, 2003).

This research has been conducted using document analysis, one of the qualitative research methods. Qualitative research methods include data collection techniques such as observation, interviews, and document analysis. This method aims to present perceptions and events in a realistic and holistic manner within their natural environments (Yıldırım & Şimşek, 2005).

Population and Sample

The population of the study consists of software programs used in music production and the computers that operate them, as well as all analog-based processors. The sample, on the other hand, includes tone-frequency-dynamics-based software processors operating within a Digital Audio Workstation (DAW), alongside analog-based processors.

Data Collection Tool

Prior to data collection, a two-channel mono recording consisting of vocal and guitar—two of the most commonly used elements in the music industry—was prepared. The vocal channel is 32 seconds long and the guitar channel is 19 seconds long. Both were imported as two separate audio tracks into Steinberg's Cubase 12 digital audio workstation.

To compare with software-based signal processors, the vocal and guitar recordings were processed using the Universal Audio Teletronix LA-2A Classic Leveling Amplifier, an analog signal processor located at Babajim Studios in Istanbul.

Conversely, to compare with analog-based signal processors, the same vocal and guitar recordings were processed using the following software-based signal processors: UAD Teletronix LA-2A, IK Multimedia White 2A Leveling Amplifier, Cakewalk CA-2A T-Type Leveling Amplifier, Overloud Comp LA, and Waves CLA-2A Classic Compressors.

Figure 1. *Babajim İstanbul Studio*

All visual and auditory applications involving the analog-based and software-based signal processors in this study were carried out within Cubase 12. During the creation of both the visual and auditory applications of the audio files, Apollo x16 A/D and D/A converters by Universal Audio were used. The entire process of visual and auditory application of the audio files was also executed through this audio interface.

In order to maintain consistency in the sound source parameters, the same audio loops were used throughout all recording stages, and volume levels were consistently kept at 0 dB to ensure balance.

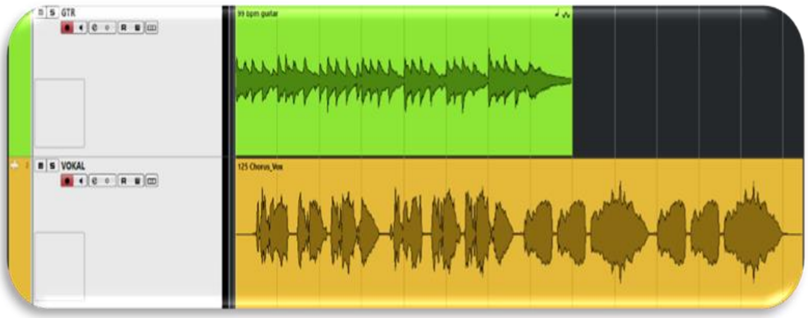

Figure 2. *Guitar and Vocals Wiew in DAW*

Figure 3. *DAW Mix Settings For Guitar and Vocals*

During the processing of sound sources, the audio files used within the DAW were routed through a digital converter, sent out, and then re-routed back in through designated analog devices via a patchbay. Throughout this process, the audio level was consistently maintained and re-recorded. The audio files were processed through the analog devices following an input-output sequence within this loop.

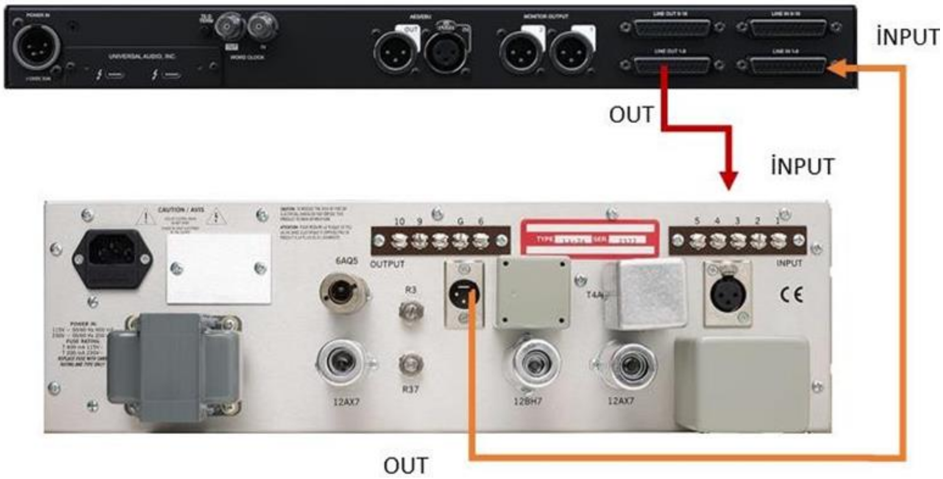

Figure 4. *Signal Flow Used*

In this study, pre-recorded vocal and guitar tracks were utilized. These recordings, captured using industry-standard microphones and analog equipment, possess measurable qualities that enable objective analysis through various audio analysis software. Consequently, the use of these recordings within the analysis process was deemed appropriate. The vocal and guitar loops were imported into Cubase 12, a digital audio workstation developed by Steinberg, and assigned to two separate audio tracks. The first track contains the audio file previously processed through analog hardware, while the second track includes the identical performance, this time processed solely with software-based tools.

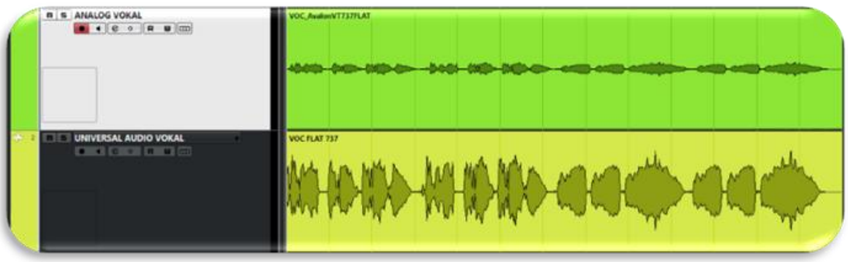

Figure 5. *Image Of Signal In Daw*

The prepared loops consist of a 28-second vocal track and a 26-second guitar track. Frequency analysis of both the analog and software-based versions of the analog sound recordings was conducted using the Fabfilter Pro-Q3 equalizer spectrum analyzer, an industrial software, over a 30-day trial period. During this process, no filtering was applied on the equalizer, as the goal was solely to obtain spectrum analysis visuals. The Pro-Q3 spectrum frequency analyzer was used for all sound recording files, including both analog and their digital derivatives.

Figure 6. I*mage Of Fabfilter Pro-Q3*

In this study, two different channel loops, vocal and guitar, were adjusted to a 0 dB (zero decibel) level. The noise levels and dynamic characteristics before this level were measured using "mvMeter 2," a free plugin developed by TB Pro Audio. Both channels, vocal and guitar, were exported as mono audio files in "wav" format, with a sample rate of 44,100 Hz and a bit depth of 16, from a recording station. The exported audio files were then transferred back into digital audio workstation (DAW) software, where the sound and frequency values were analyzed in detail. The master output of the channels was fixed at the 0 dB level, and the same noise level values were calculated for both analog and digital software.

Figure 7. *Image Of TB Pro Audio mvMeter-2*

In the study, the Insight 2 Pro Metering tool from iZotope was used for multiple visualizations of the stereo field activity and its previous events. In this study, stereo correlation measurements between the left and right channels were performed. The analyses of the processed audio files derived from both analog and their corresponding software versions were conducted using the vector scope graph and field measurements of the Insight 2 Pro meter. This method allowed for a more detailed examination of the dynamics of the stereo field and the relationship between the channels.

Figure 8. *Image Of Insight 2 Pro Metering Vektorskop*

In the creation of real-time three-dimensional spectrogram graphics, the Insight 2 Pro spectrogram tool from iZotope was used, and detailed topographic sound maps were obtained. In these sound maps, the changes and progress of the signal's frequency domain content over time were displayed with three-dimensional visual graphics, presenting the static, dynamic, and natural timbres of the sounds visually. With the help of spectrogram graphics, comparisons were made between all the audio recording files of analog and their corresponding analog equipment simulations. The purpose of these examinations was to reveal the differences between analog hardware equipment and their exact software simulations, and to analyze these differences in more detail.

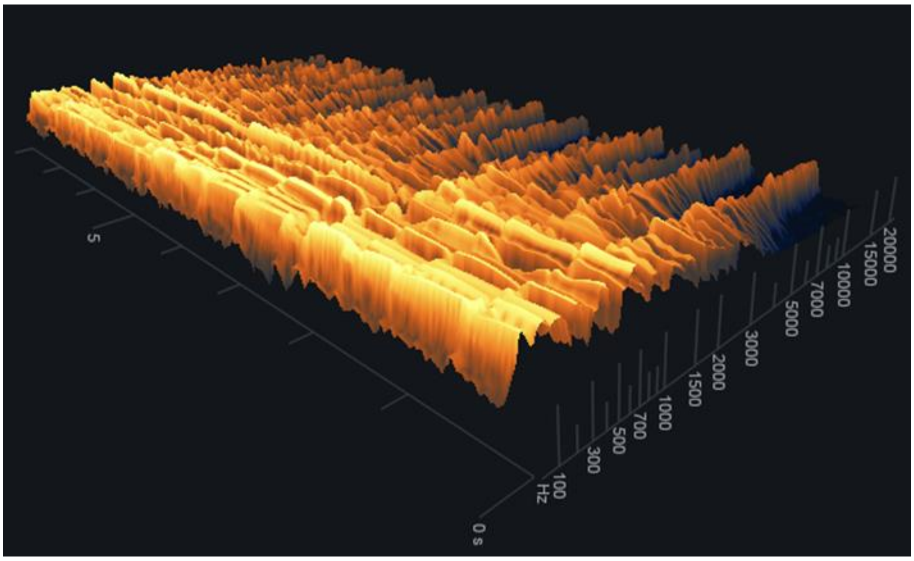

Figure 9. *Image Of Insight 2 Pro Spektrogram 3D*

In the processing of sound sources, the audio files used within the DAW (Digital Audio Workstation) were output through a digital converter and re-input through selected analog devices, with the sound balance being consistently recorded again. The audio files were passed through the analog devices in an "out-input-out-input" sequence within this loop. In the study, Steinberg's Cubase 12 Digital Audio Workstation (DAW) was used in the processing of sound sources and the creation of sound analyses. The visual and auditory applications of the analog signal processors and digital signal processors, which form the basis of the study, were entirely carried out on this software. For the creation of the visual and auditory applications of the audio files, Universal Audio's Apollo x16 A/D and D/A conversion devices were used. All the visual and auditory processes of the audio files were carried out on this sound card.

FINDINGS AND COMMENTS

In the findings and discussions, analog-based signal processors were compared with the software-based plug-ins of the same analog devices. As a result of these comparisons, the differences between analog-based signal processors and software-based plug-ins were revealed.

Figure 10. *Teletronix LA-2A ve Over Loud, T-Racks, Waves, Cakewalk ve Universal Audio Software*

For the guitar and vocal channels, the limiter parameters of the Teletronix LA-2A analog device were set to Gain: 40, Peak Reduction: 60. Similarly, the Over Loud, T-Racks, Waves, Cakewalk, and Universal Audio software were set with the same parameters as the analog device, with Gain: 40 and Peak Reduction: 60.

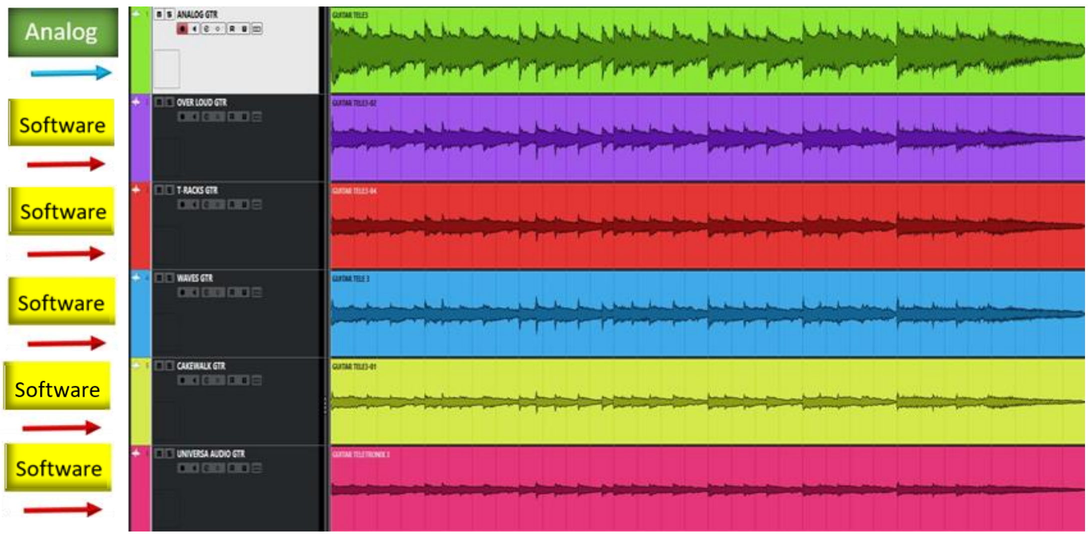

Figure 11. *Guitar View In DAW*

In the comparison, after the data obtained from the signal processors, the signal waveforms of the guitar within the DAW are shown in Figure 11. The first channel displays the signal passing through the analog processors. The second channel shows the Over Loud, the third channel shows T-Racks, the fourth channel shows Waves, the fifth channel shows Cakewalk, and the sixth channel shows the signal waveform of the software-based processor from Universal Audio. As seen from the guitar's waveform within the DAW, the dynamic range is ordered from high to low from the first channel to the sixth channel.

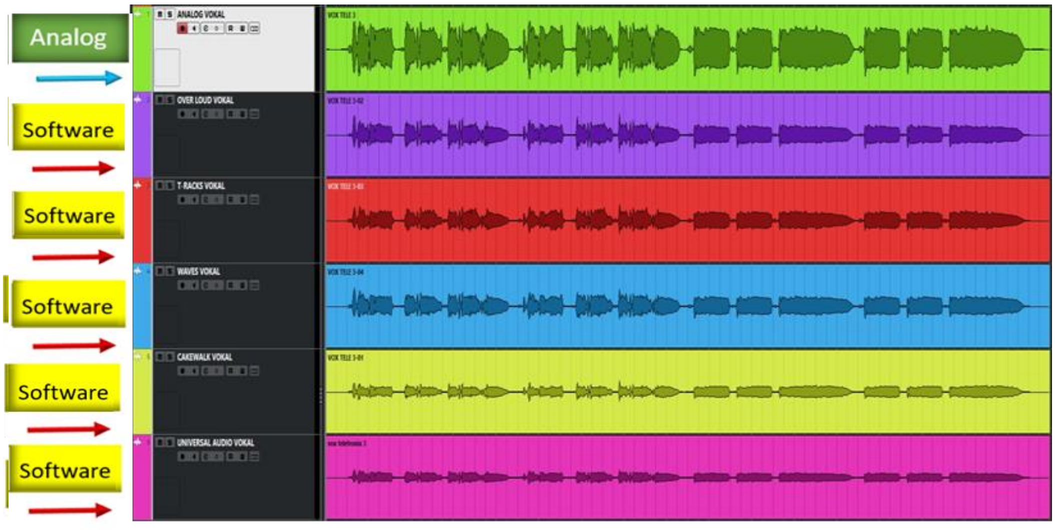

Figure 12. *Vocal View In DAW*

With the same settings, the signal waveforms of the vocal within the DAW, after the data obtained from the signal processors, are shown in Figure 12. The first channel displays the signal passing through the analog processors. The second channel shows the Over Loud, the third channel shows T-Racks, the fourth channel shows Waves, the fifth channel shows Cakewalk, and the sixth channel shows the signal waveform of the software-based processor from Universal Audio. As seen from the vocal's waveform within the DAW, the dynamic range is ordered from high to low from the first channel to the sixth channel.

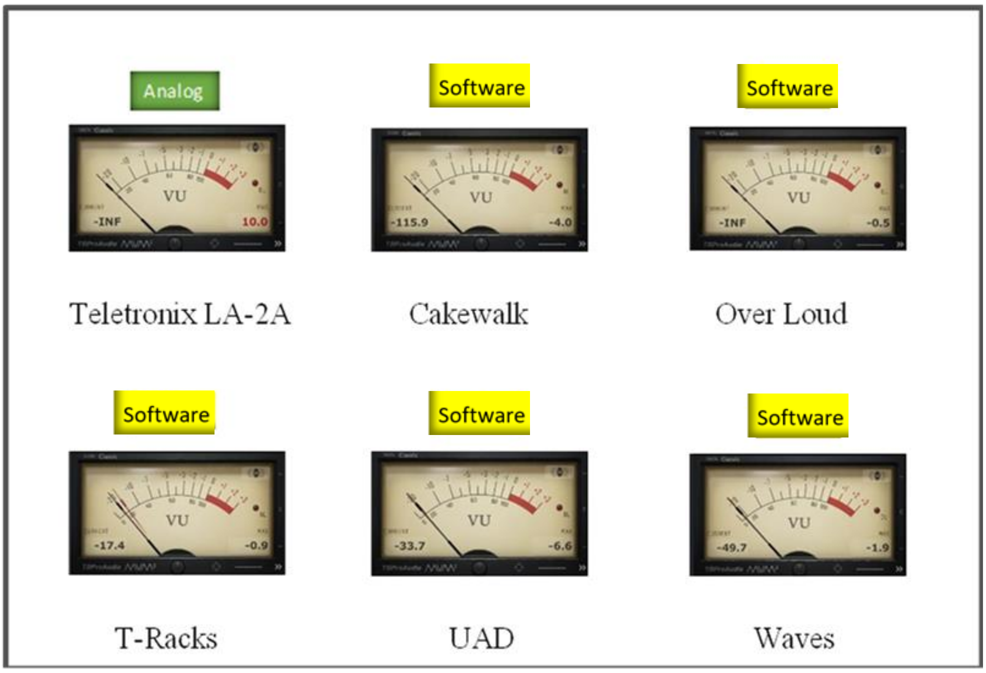

Figure 13. *Guitar's mvMeter-2 Maximum Peak Image*

After the data obtained from the Teletronix LA-2A analog device and software-based signal processors from Over Loud, T-Racks, Waves, Cakewalk, and UAD companies, the maximum peak level of the master output fader in the guitar channel in dB is shown in Figure 13 on the mvMeter-2.

In this figure, the maximum peak level of the data obtained from the Teletronix LA-2A analog device for the guitar channel, after compression, was determined to be 10 dB with a VU meter measurement. On the other hand, the data obtained from the analog device and the software-based signal processors from Over Loud, T-Racks, Waves, Cakewalk, and UAD companies resulted in different dB levels on the VU meter for each software:

- **Teletronix LA-2A:** 10 dB (analog)
- **Over Loud:** -0.5 dB (software)
- **T-Racks:** -0.9 dB (software)
- **Waves:** -1.9 dB (software)
- **Cakewalk:** -4.0 dB (software)
- **UAD:** -6.6 dB (software)

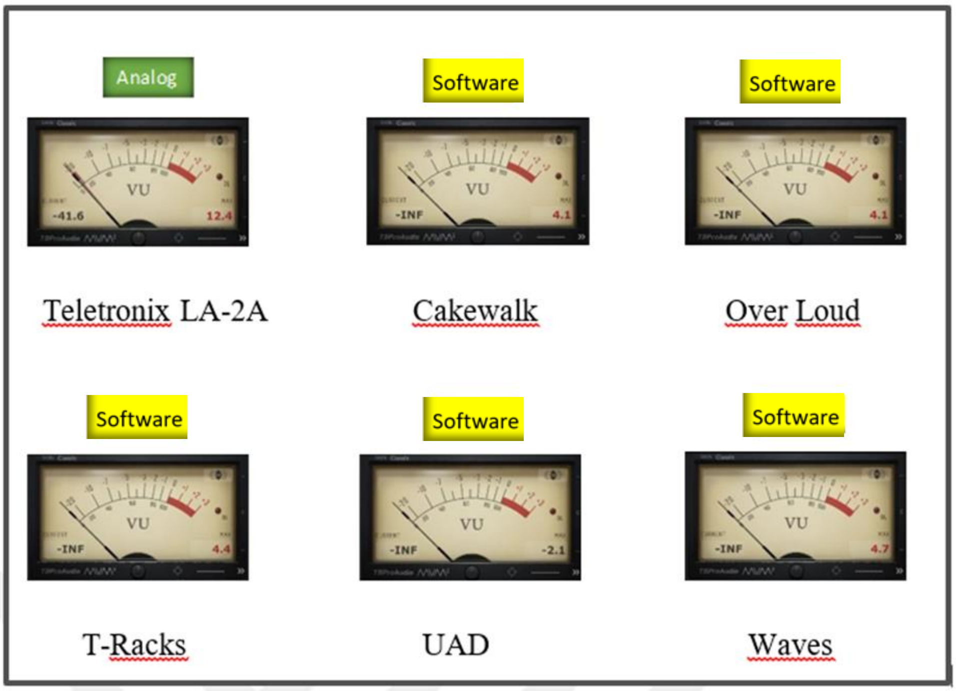

Figure 14. *Vocal's mvMeter-2 Maximum Peak Image*

After the data obtained from the Teletronix LA-2A analog device and software-based signal processors from Over Loud, T-Racks, Waves, Cakewalk, and UAD companies, the maximum peak level of the master output fader in the vocal channel in dB is shown in Visual 14 on the mvMeter-2.

In this visual, the maximum peak level of the data obtained from the Teletronix LA-2A analog device for the vocal channel, after compression, was determined to be 12.4 dB with a VU meter measurement. On the other hand, the data obtained from the analog device and the software-based signal processors from Over Loud, T-Racks, Waves, Cakewalk, and UAD companies resulted in different dB levels on the VU meter for each software:

- **Teletronix LA-2A:** 12.4 dB (analog)
- **Over Loud:** 4.1 dB (software)
- **T-Racks:** 4.4 dB (software)
- **Waves:** 4.7 dB (software)
- **Cakewalk:** 4.1 dB (software)
- **UAD:** -2.1 dB (software)

It was observed that there are differences in the dB levels and compression ratios between the analog and the software versions of the same analog equipment, based on the data obtained from the guitar channel and the vocal channel. In the data obtained from the guitar channel, a greater difference in sound levels between the analog and software versions is observed, while in the data obtained from the vocal channel, a smaller difference in sound levels is observed.

Figure 15. *Guitar Spectrum Analysis Image (Analog)*

In Figure 15, the spectrum analysis of the guitar channel data obtained from the Teletronix LA-2A analog device is shown. According to this analysis, the guitar is observed to have frequency components in the range of approximately 100 Hz to 20 kHz. The spectrum analysis reveals that the most prominent frequency regions of the instrument are concentrated around 183 Hz, 1.2 kHz, 2.6 kHz, 12 kHz, and 22 kHz. The Teletronix LA-2A analog device is seen to control the dynamic range of the signal without applying any filtering to the lower frequency range of the guitar.

Figure 16. *Guitar Spectrum Analysis Image Cakewalk*

Figure 17. *Guitar Spectrum Analysis Image Over Loud*

Figure 18. *Guitar Spectrum Analysis Image T-Racks*

Figure 19. *Guitar Spectrum Analysis Image UAD*

Figure 20. *Guitar Spectrum Analysis Image Waves*

In Figure 16, the spectrum analysis of the guitar channel data obtained from the Cakewalk software-based signal processor is shown; in Figure 17, from Over Loud; in Figure 18, from T-Racks; in Figure 19, from UAD; and in Figure 20, from Waves. According to these analyses, all the software versions show that the guitar has frequency components in the range of approximately 100 Hz to 20 kHz, just like the data obtained from the analog device.

In the Cakewalk software, the most prominent frequency regions of the instrument are concentrated around 172 Hz, 1.2 kHz, 2.6 kHz, 12 kHz, and 22 kHz. In the Over Loud software, the prominent frequencies are observed around 172 Hz, 1.2 kHz, 2.6 kHz, 4.9 kHz, and 12 kHz. In the T-Racks software, the prominent frequencies are around 172 Hz, 1.2 kHz, 2.6 kHz, 4.9 kHz, and 12 kHz. In the UAD software, the frequencies are concentrated around 183 Hz, 1.2 kHz, 2.6 kHz, 4.8 kHz, and 12 kHz. In the Waves software, the frequencies are concentrated around 172 Hz, 1.2 kHz, 2.6 kHz, 12 kHz, and 22 kHz.

As a result of these observations, it is seen that the analog device processes all the frequencies, while the software only controls the concentrated frequency range of the guitar channel by applying a low-cut filter.

Figure 21. *Vocal Spectrum Analysis Image (Analog)*

In Figure 21, the spectrum analysis of the vocal channel data obtained from the Teletronix LA-2A analog device is shown. According to this analysis, the vocal has frequency components in the range of approximately 200 Hz to 20 kHz. The spectrum analysis reveals that the most prominent frequency regions of the vocal are concentrated around 420 Hz, 1.7 kHz, 2.9 kHz, 10 kHz, and 20 kHz. The Teletronix LA-2A analog device is seen to control the dynamic range of the signal without applying any filtering to the lower frequency range of the vocal.

In Figure 22, the spectrum analysis of the vocal channel data obtained from the Cakewalk software-based signal processor is shown; in Figure 23, from Over Loud; in Figure 24, from T-Racks; in Figure 25, from UAD; and in Figure 26, from Waves. According to these analyses, all the software versions show that the vocal has frequency components in the range of approximately 200 Hz to 20 kHz, just like the data obtained from the analog device.

In the Cakewalk software, the most prominent frequency regions of the vocal are concentrated around 420 Hz, 1.7 kHz, 3 kHz, 8.1 kHz, and 17 kHz. In the Over Loud software, the prominent frequencies are observed around 420 Hz, 1.7 kHz, 3 kHz, 9.6 kHz, and 17 kHz. In the T-Racks software, the prominent frequencies are concentrated around 378 Hz, 1.7 kHz, 2.9 kHz, 10 kHz, and 18 kHz. In the UAD software, the frequencies are observed around 420 Hz, 1.7 kHz, 3 kHz, 9.6 kHz, and 20 kHz. In the Waves software, the frequencies are concentrated around 420 Hz, 1.7 kHz, 2.9 kHz, 6.9 kHz, and 13 kHz.

As a result of these observations, it is seen that the analog device processes all the frequencies, while the software only controls the concentrated frequency range of the vocal channel by applying a low-cut filter.

Figure 22. *Vocal Spectrum Analysis Image Cakewalk*

Figure 23. *Vocal Spectrum Analysis Image Over Loud*

Figure 24. *Vocal Spectrum Analysis Image T-Racks*

Figure 25. *Vocal Spectrum Analysis Image UAD*

Figure 26. *Vocal Spectrum Analysis Image Waves*

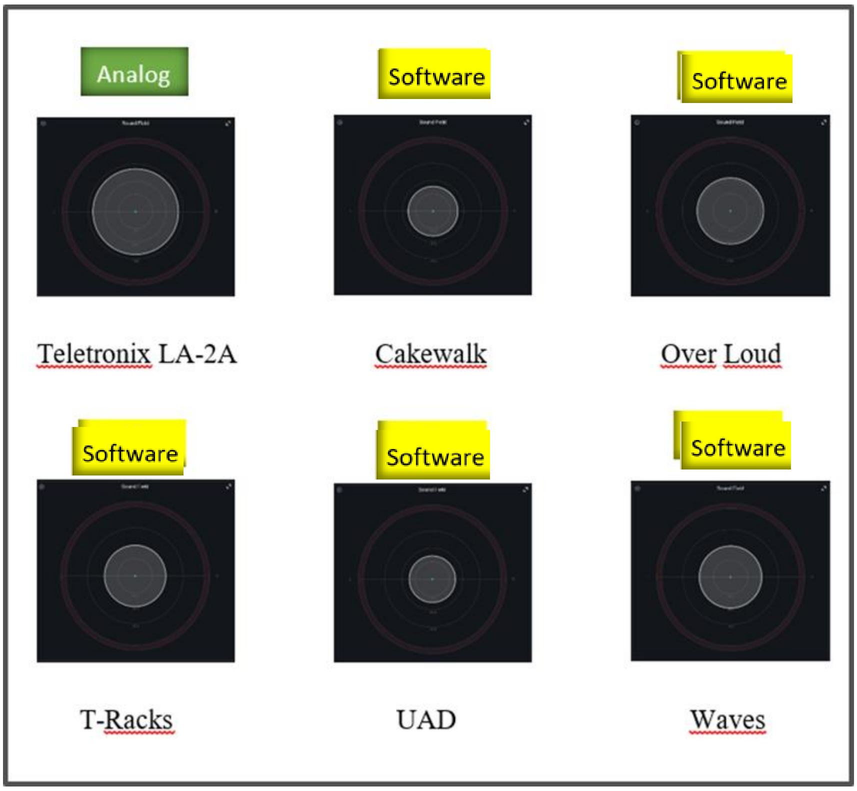

Figure 27. *Guitar Vektorskop Image*

In Figure 27, the vector scope graph area measurements of the guitar channel data obtained from the Teletronix LA-2A analog device, and its software versions are shown. The maximum stereo field activity of the guitar is demonstrated, with stereo correlation measurements made between the left and right channels. It is observed that the analog device spreads the guitar sound over a wider area compared to the software-based signal processors in a 360-degree surround distribution. Additionally, it was observed that in the software-based signal processors, the ratios and values of the stereo field activity differ from one another.

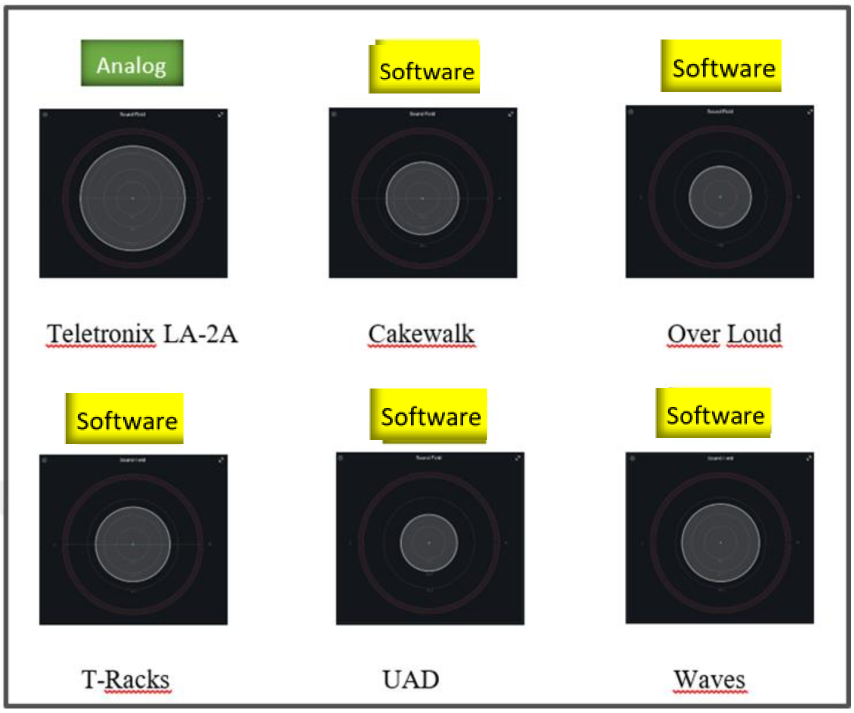

Figure 28. *Vocal Vektorskop Image*

In Figure 28, the vector scope graph area measurements of the vocal channel data obtained from the Teletronix LA-2A analog device and its software versions are shown. The maximum stereo field activity of the vocal is demonstrated, with stereo correlation measurements made between the left and right channels.

Figure 28 also shows the vector scope comparison graph of the vocal channel data obtained from the Teletronix LA-2A analog device and its software versions. It is observed that the analog device spreads the guitar sound over a wider area compared to the software-based signal processors in a 360-degree surround distribution. Additionally, it was observed that in the software-based signal processors, the ratios and values of the stereo field activity differ from one another.

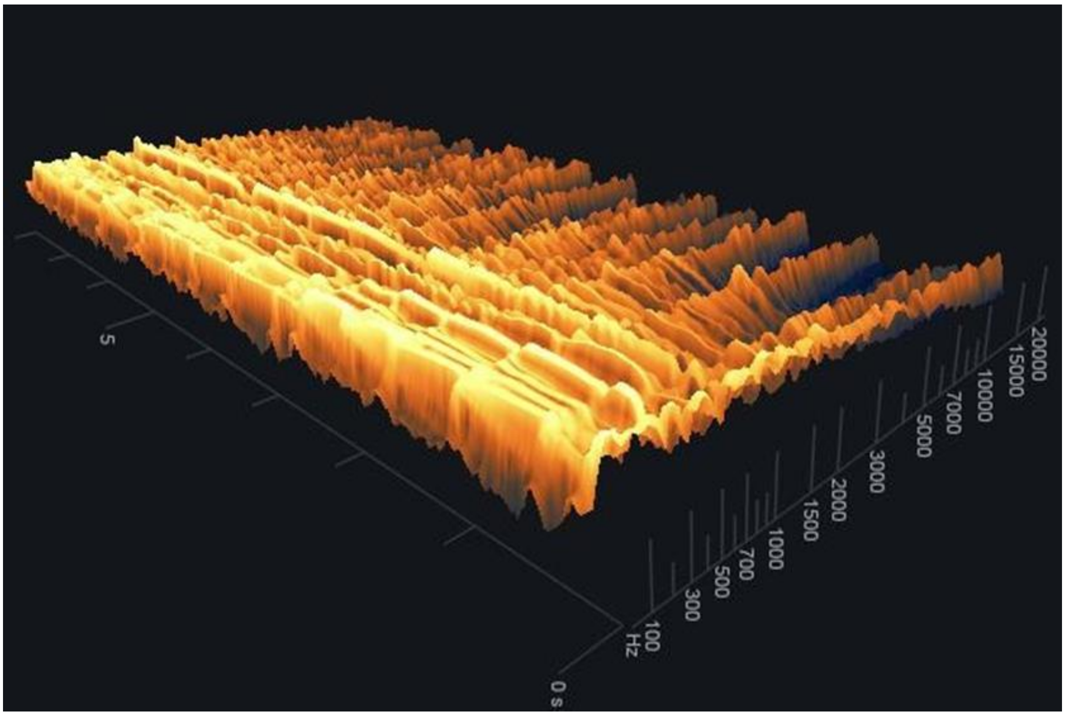

Figure 29. *Spectrogram Graph Of Guitar (Analog)*

The 3D spectrogram graph of the guitar channel data obtained from the Teletronix LA-2A analog device is shown in Figure 29. In this analysis, a detailed topographic sound map of the guitar has been created. In the spectrogram graph of the guitar in Visual 29, the signal's static, dynamic, and natural tone is displayed in the expected graphical shape over time after processing with the Teletronix LA-2A analog device.

In Visual 30, the 3D spectrogram graph of the guitar channel data obtained from the Cakewalk software, in Visual 31 from Over Loud, in Visual 32 from T-Racks, in Visual 33 from UAD, and in Visual 34 from Waves are shown. In the spectrogram graph of the analog device in Visual 29, the signal's frequency concentration is seen to focus between 100 Hz and 3000 Hz. In contrast, the data from the software-based processors show that the signal's frequency energy decreases between 100 Hz and 3000 Hz, and the energy of the frequency between 3000 Hz and 20000 Hz is lost.

In the spectrogram graph of the analog device in Visual 29, the saturation value is shown to be at the expected level. In Visual 30, the saturation value in the Cakewalk software, in Visual 31 in Over Loud, in Visual 32 in T-Racks, in Visual 33 in UAD, and in Visual 34 in Waves software-based signal processors is observed to be at a lower temperature level.

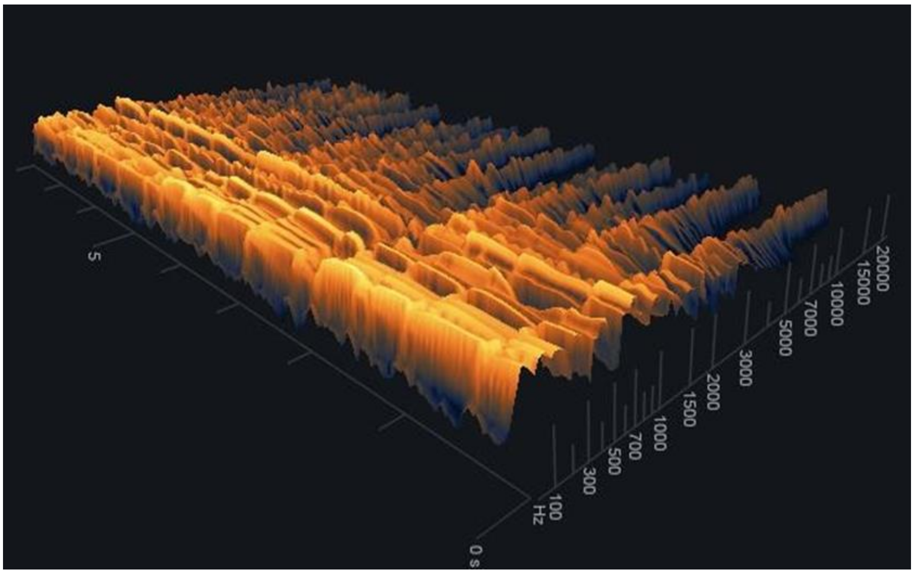

Figure 30. *Spectrogram Graph Of Guitar Cakewalk*

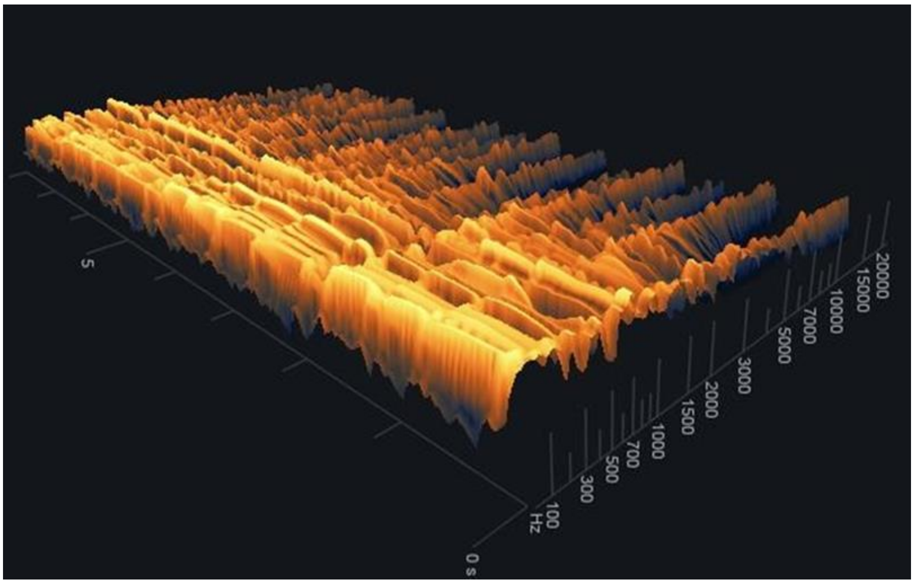

Figure 31. *Spectrogram Graph Of Guitar Over Loud*

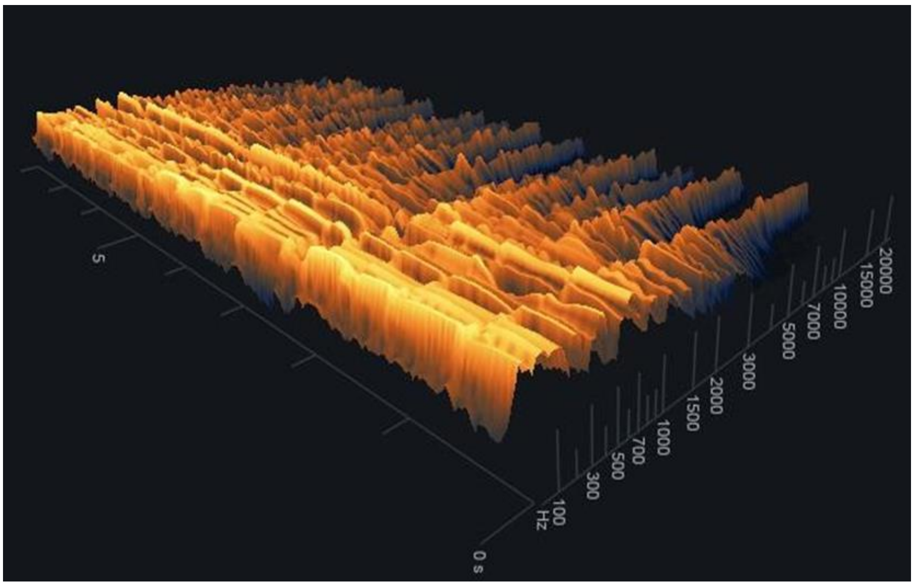

Figure 32. *Spectrogram Graph Of Guitar T-Racks*

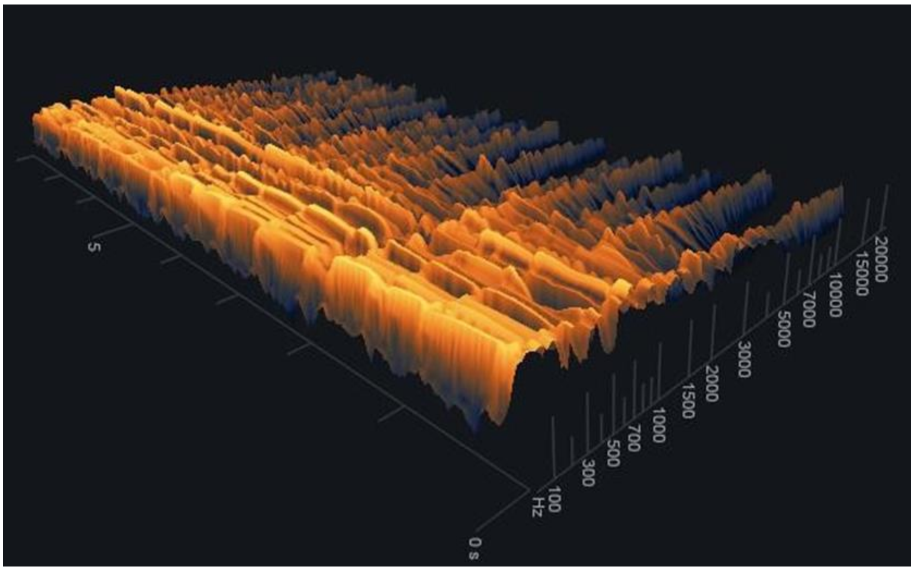

Figure 33. *Spectrogram Graph of Guitar UAD*

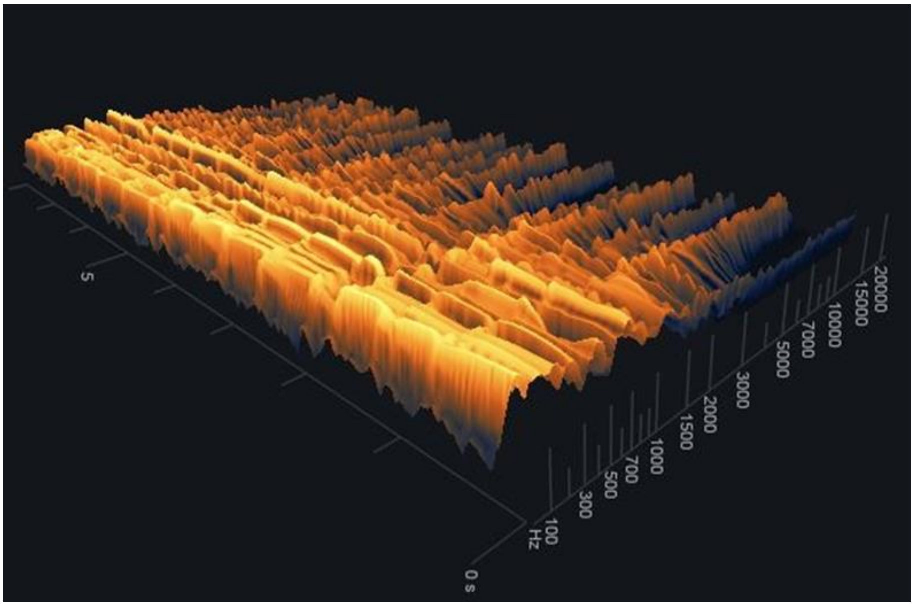

Figure 34. *Spectrogram Graph of Guitar Waves*

The 3D spectrogram graph of the vocal channel data obtained from the Teletronix LA-2A analog device is shown in Figure 35. In this analysis, a detailed topographic sound map of the vocal has been created. In the spectrogram graph of the vocal in Figure 35, the signal's static, dynamic, and natural tone is displayed in the expected graphical shape over time after processing with the Teletronix LA-2A analog device.

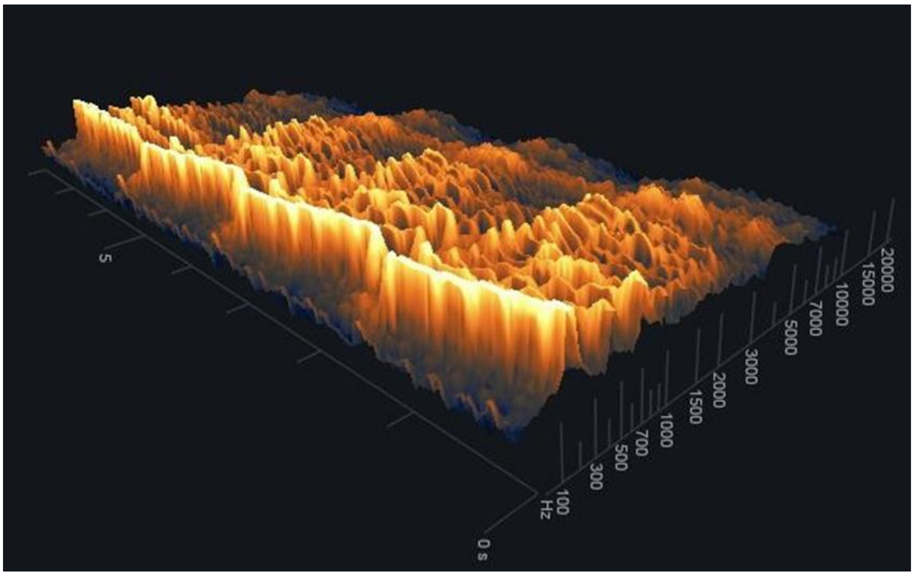

Figure 35. *Spectrogram Graph of Vocal (Analog)*

In Figure 36, the 3D spectrogram graph of the vocal channel data obtained from the Cakewalk software, in Figure 37 from Over Loud, in Figure 38 from T-Racks, in Figure 39 from UAD, and in Figure 40 from Waves software-based signal processors are shown. In the spectrogram graph of the analog device in Figure 35, the signal's frequency concentration is seen to focus between 100 Hz and 5000 Hz, while the data obtained from the software-based processors show that the signal's frequency energy decreases between 100 Hz and 3000 Hz, and the frequency energy between 3000 Hz and 20000 Hz is lost.

In the spectrogram graph of the analog device in Figure 35, the saturation value is shown to be at the expected level. In Figure 36 (Cakewalk), Figure 37 (Over Loud), Figure 38 (T-Racks), Figure 39 (UAD), and Figure 40 (Waves), the saturation values of the software-based signal processors are seen to be at a lower temperature level.

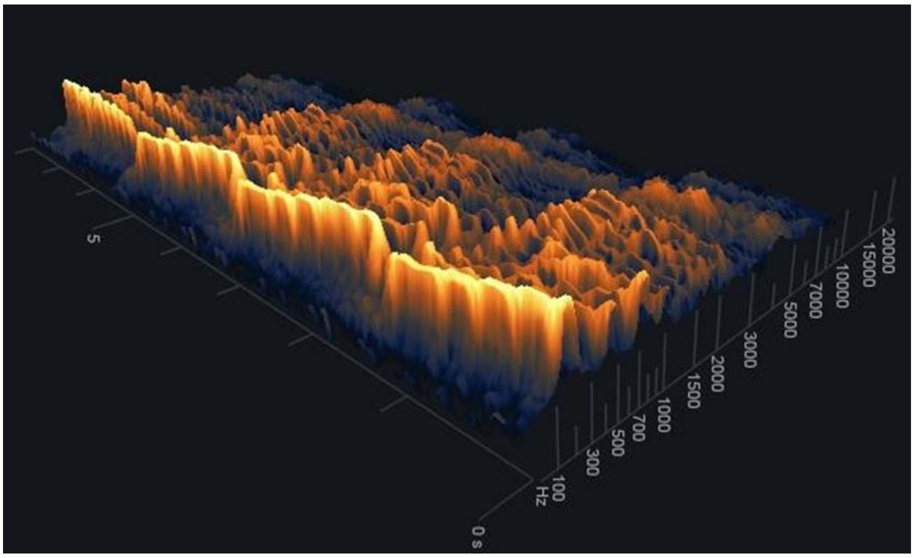

Figure 36. *Spectrogram Graph of Vocal Cakewalk*

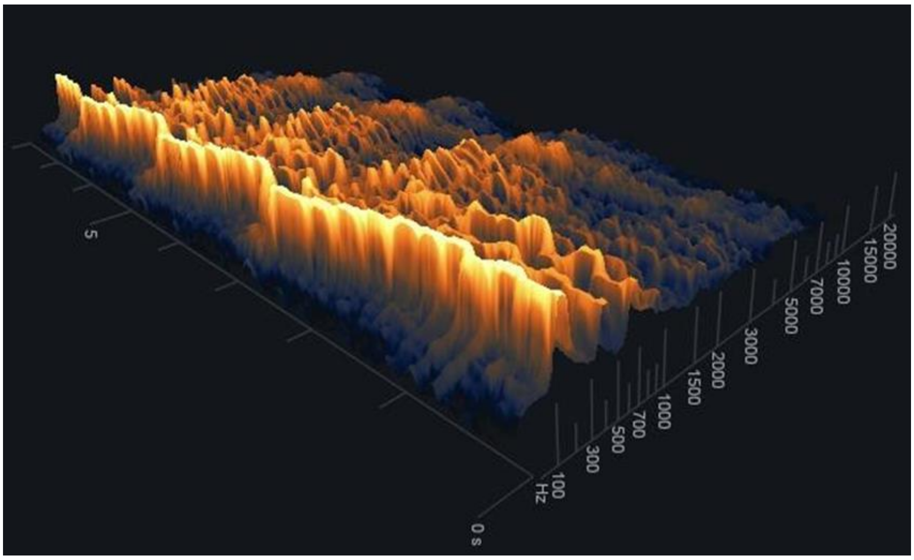

Figure 37. *Spectrogram Graph of Vocal Over Loud*

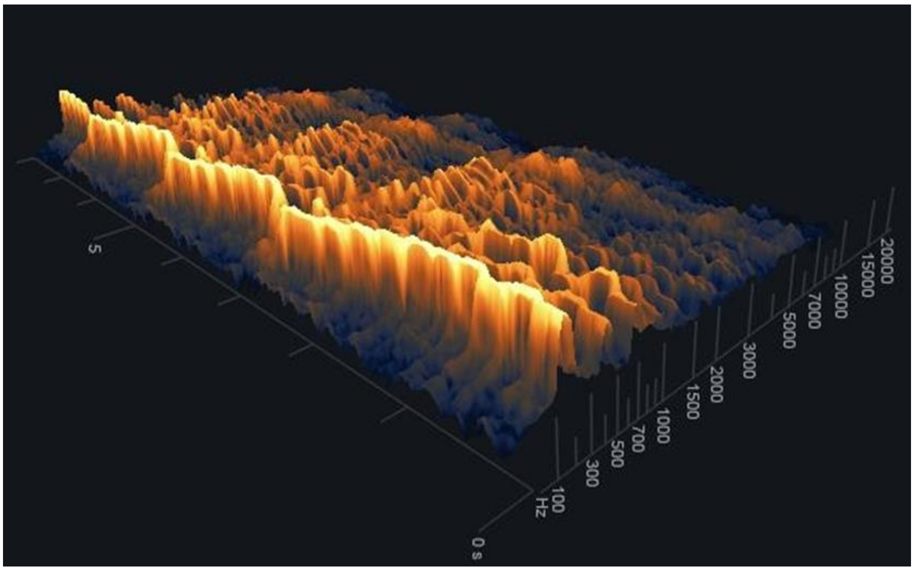

Figure 38. *Spectrogram Graph of Vocal T-Racks*

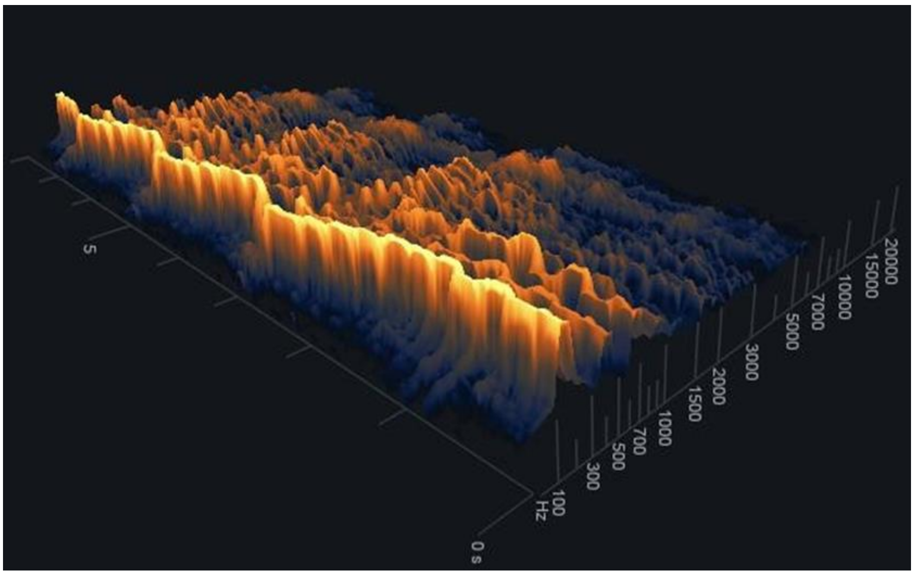

Figure 39. *Spectrogram Graph of Vocal UAD*

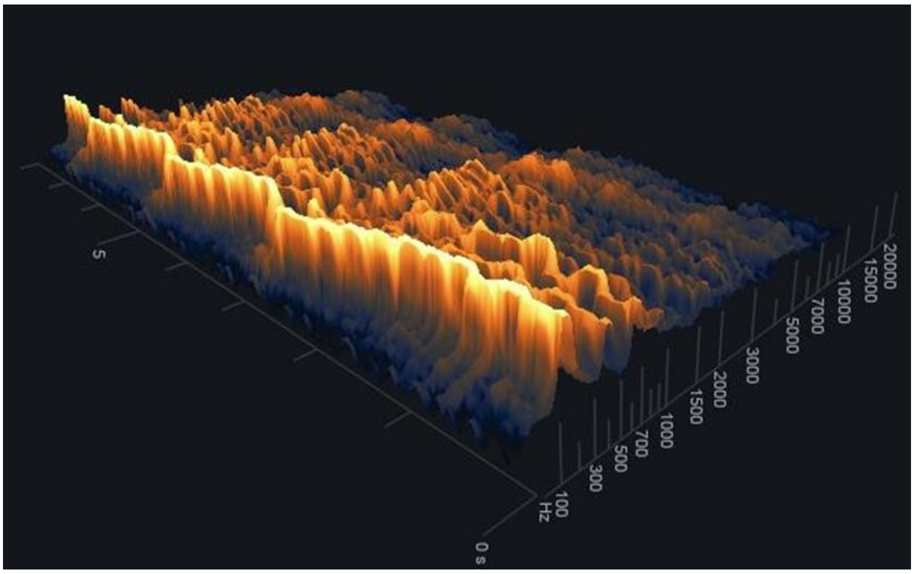

Figure 40. *Spectrogram Graph of Vocal Waves*

CONCLUSION, DISCUSSION AND RECOMMENDATIONS

When comparing the guitar and vocal signals obtained from analog-based signal processors and software-based processors during the mixing stage of the recording process, it was concluded that software-based processors do not replicate the harmonic coherence provided by analog signal processors. In the guitar signals processed through analog signal processors, the master output peak levels measured with a VU meter were observed to be higher in dB compared to those processed with software-based processors. A similar situation was also observed in vocal signals; analog signal processors showed higher peak levels, whereas software-based processors remained at lower levels.

Spectrum analyses revealed that the frequency components of analog signal processors do not match the energy ranges of software-based processors. While analog processors do not apply low-cut filtering on the lower frequency range, software-based processors automatically apply cuts between 20 Hz and 50 Hz. It was concluded that in mix sessions involving multiple plugin chains, software-based processors apply significant cuts in the 20–50 Hz range, effectively muting those frequencies. This indicates that analog and software-based processors do not process frequency content with the same energy levels. While frequency ratios in analog processors remain stable, those in software-based processors vary.

Furthermore, software-based processors were found to be unable to fully replicate the RMS levels of analog processors, as they alter the dynamic range and noise floor of the signal. The saturation values of analog signal processors fluctuate during operation, whereas those in software-based processors remain constant. Software processors do not emulate the temperature-dependent saturation behavior seen in analog hardware, and they do not experience heat-related effects. Therefore, compression, frequency processing, and saturation levels remain fixed in software processors.

In the spectrogram graphs, which show how the spectral content of the signal evolves over time, it was observed that the frequency intensity in analog signal processors generally concentrated between 100 Hz and 2000 Hz. In contrast, signals processed by software-based processors showed reduced energy in the 100–2000 Hz range, and energy loss was observed in the 5000–20000 Hz range. This indicates that the saturation characteristics of analog and software processors differ in terms of heat and intensity.

The development of signal processing technologies in audio engineering and music production represents a significant milestone in the evolution of recording and mixing processes. The choice between analog and software-based signal processors is not only based on technical requirements but also on economic and operational considerations. This study compares the advantages and limitations of both systems in light of studio budgets, physical space limitations, and opportunities for technology-based development.

Analog signal processors are generally more costly due to their design. Their physical structure and required hardware components increase their overall expense and pose challenges in terms of portability and studio integration. Their usage in studios often presents spatial constraints and difficulties in transport and setup.

On the other hand, software-based signal processors offer a more economical solution as they operate on computer-based digital platforms. These systems, being integrated into the digital environment, are free from issues related to physical placement and portability. Additionally, software-based processors are more flexible and accessible, making them a more efficient option, especially in terms of studio budgets and space limitations.

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