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Comparison of Analog and Software-Based Processors in the Mixing Processabstract

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Abstract

This research aims to comparatively examine the technical and auditory effects of analog-based and software-based processors used in the mixing process. The study targets evaluating the performance of software processors running on digital audio workstations (DAWs), which are widely used in modern music production, against physical analog devices during mixing. The research was conducted using a descriptive qualitative approach and document analysis method. The research universe consists of analog and software-based signal processors, while the sample comprises DAW-based mixing projects utilizing these processors. Within the scope of the study, both processor types were compared under similar conditions; level balances, input gains, and harmonic contents were standardized for evaluation. The findings reveal that analog processors provide a more natural harmonic richness and warmth in frequency characteristics, whereas software-based processors simulate these characteristics to some extent but sometimes yield a linear and fixed tonal outcome. Additionally, the research shows that software-based processors offer advantages such as ease of use and reproducibility, while analog devices impart a more organic effect on the sound's dynamic and tonal character. In conclusion, the study assesses the role of both processor types in the mixing process from technical and aesthetic perspectives and provides an analysis capable of guiding user preferences.

Keywords Plug-in · Analog Digital Signal Processor (DSP) · Music Technologies

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Comparison of Analog and Software-Based Processors in the Mixing Processabstract

The origins of sound recording technology date back to the early 19th century. From Thomas Young's acoustic experiments in 1807 to Léon Scott de Martinville's invention of the phonautograph in 1857-which could visually record sound waves-and Thomas Edison's 1877 phonograph that enabled both recording and playback of sound, the foundations of sound technology were laid through significant innovations (Cumhuriyet Bilim Teknik, 1994; CNN Türk, 2008). In 1889, Valdemar Poulsen developed the "telegraphone," the first magnetic recording device, introducing electromagnetic systems into the process of sound recording (Mumma, Rye & Kernfeld, 2010). Throughout the 20th century, institutions such as Bell Laboratories, Western Electric, and the Victor Talking Machine Company advanced the development of microphones, recorders, and electromechanical systems, allowing for clearer and more controllable sound capture (Burgess, 2014; Schmidt-Horning, 2013).

During the 1930s, the development of dynamic processors such as compressors, limiters, and expanders added an aesthetic layer to the recording process beyond mere technical functionality (Izhaki, 2008; Öcek, 2010; Pasinlioğlu, 2019). In the 1950s, the introduction of multitrack recording systems by Les Paul and Ampex laid the groundwork for modern mixing practices by enabling independent recording and processing of sound layers (Önen, 2010). These developments led to the separation of recording and mixing processes, each with its own set of technical requirements.

Analog hardware devices were central to music production particularly during the 1960s and 1970s. Their characteristic warmth, saturation, and harmonic richness added distinct tonal qualities to audio recordings. However, with the rapid rise of digital technologies in the late 1990s, software-based plug-ins began to emulate the functions of analog devices and gained widespread adoption in music production (Huber, 2010; Rumsey, 1994). Owing to their affordability, ease of use, instant accessibility, and flexible automation capabilities, digital processors became especially popular in home studios (Lefford, 2000). This shift has given rise to an ongoing debate surrounding analog and digital technologies, both from technical and perceptual perspectives.

Various comparative studies in the literature have examined the two technologies from different angles. Smith (2015) highlighted the perceived warmth and harmonic characteristics of analog devices, while Johnson and Lee (2018) analyzed their frequency response, dynamic range management, and distortion profiles from a technical standpoint. Williams (2017) assessed the accuracy of digital emulations, and Chen et al. (2020) compared the perceptual effects of analog and digital mixing through listening tests. Collectively, these studies suggest that analog processors offer distinct coloration and aesthetic character, whereas digital processors provide cleaner, more flexible, and reproducible signal processing.

This study focuses not only on the historical development of sound recording technology but more specifically on the mixing process, examining the technical and perceptual distinctions between analog hardware and software-based processors. The analog-versus-digital debate is not merely a technological preference but a multidimensional decision involving aesthetic perception, usability, and production workflows. Therefore, this study evaluates both approaches through metrics such as frequency response, dynamic range, harmonic distortion, sonic coloration, ease of use, automation potential, and cost. Through a document analysis method involving technical literature, professional experiences, and practical applications, this research aims to provide a comparative framework that enables mixing engineers to make more informed decisions (Karasar, 2003; Yıldırım & Şimşek, 2005). Thus, the analog-digital discussion is addressed not only in terms of technology but also through its implications for auditory aesthetics and production practices.

Problem Situation

The rapid advancement of technology has fundamentally transformed album production processes in home studios. This transformation has accelerated with the adaptation of analog devices to digital platforms and the development of digital processors. Digital processors have become increasingly preferred due to their cost-effectiveness and user-friendly features compared to analog equipment. This situation has created a significant choice between analog and software-based devices influenced by the digitalization of music production processes. The aim of this study is to reveal the technical and technological differences between analog and software-based signal processors commonly used in the mixing stage. Thus, it aims to enable mixing engineers and music producers to make informed decisions based on the effects of these two processor types on frequency response, dynamic control, harmonic distortion, and auditory character.

Problem Statement

What are the differences between analog-based processors and software-based processors during the mixing process?

Significance of the Study

A comparative analysis of analog and digital (software-based) processors in the mixing process-based on sound quality, processing performance, control flexibility, cost-effectiveness, technological compatibility, and sonic characteristics-is critically important for assessing their suitability across different mixing scenarios. Such comprehensive evaluations function as a guide for audio technology professionals, musicians, producers, and academic researchers, enabling them to make informed choices regarding which systems best meet their needs during the mixing stage. Moreover, such studies provide a solid foundation for the development of mixing-oriented industry standards and the creation of innovative solutions. Audio engineers and arrangers can achieve more effective and efficient results in their projects by leveraging the best performance from both types of processors.

Limitations of the Study

This study is limited to digital audio processing stations and their subcomponents.

Assumptions

It is assumed that during the research process, the quantitative data obtained through the spectrum analysis of the signal processors used in the production phase will accurately reveal the actual performance characteristics of both analog and software-based processors.

Methods

Research Model

This study is a descriptive qualitative research designed to compare the technical and auditory effects of analog and software-based processors used in the mixing stage of music production. In the study, both types of processors were evaluated under similar mixing conditions; level balances, input gains, and harmonic contents were stabilized to conduct a systematic analysis.

The research was carried out using the document analysis technique, one of the qualitative research methods. Within this scope, technical data, audio samples, visual documents (screenshots, setting diagrams, etc.), and literature reviews related to analog and software-based signal processors used in different mixing projects were taken as the basis.

Document analysis involves systematically examining relevant documents to collect and interpret data (Yıldırım & Şimşek, 2005). Through this method, frequency responses, harmonic distortion values, dynamic range responses, and perceptual effects on the mix of the processors were comparatively evaluated.

Population and Sample

The population of the research consists of software programs hosting music software and computers that work in harmony with these programs, along with all analog-based processors. The sample, however, consists of tone, frequency, and dynamic software-based processors, as well as analog-based processors that operate in a DAW (digital audio workstation) environment.

Data Collection Tool of the Research

Prior to the data collection phase of the study, two mono-format channels were recorded targeting the most commonly used sources in the music industry: vocals and guitar. The vocal channel was recorded for 32 seconds, and the guitar channel for 19 seconds, then transferred as two separate audio tracks into Steinberg's Cubase 12 digital audio workstation.

For comparison with software-based signal processors, the vocal and guitar recordings were processed using analog signal processors such as the Manley Passive EQ, Manley Variable Mu compressor, and Millennia Music TCL-2 at Artline Ankara Studios, and subsequently routed through a Neve 8816 Summing mixer (Figure 1).

Other recordings prepared for comparison with analog signal processors were processed via software-based processing using Universal Audio DSP-based interfaces and plugins from UAD and Plugin Alliance, then also routed through the Neve 8816 Summing mixer.

Throughout all recording and processing stages, audio levels were maintained at a constant 0 dBFS (decibels relative to full scale) in the digital domain. Gain staging principles were applied in the signal flow, and input levels were equalized for both processing chains. Faders in the DAW were kept at unity gain position (0.0 dB); however, since this alone does not guarantee auditory equivalence, the input gains of signals prior to processing were carefully equalized and standardized.

Additionally, all analog devices used were pre-calibrated according to manufacturer-specified reference levels and frequency characteristics. During calibration, a 1 kHz 0 dBFS test signal was employed, and the output responses of the devices were measured and stabilized. Software-based plugins were tested using factory default parameters, and their digital gain curves were evaluated as reference points.

Figure 1

Studio Artline (Ankara)

In the study, the visual and auditory applications of analog-based and software-based signal processors were entirely conducted within the Steinberg Cubase 12 digital audio workstation (DAW) environment (Figure 2).

Throughout all recording and analysis processes, the Universal Audio Apollo x16 A/D and D/A conversion device was used. The conversion of signals from digital to analog and back to digital was performed through this high-resolution audio interface, thereby minimizing potential quality loss within the signal chain.

To maintain constant parameters of the sound source, identical audio loops were used consistently during all recording and processing stages; each audio track was normalized to 0 dBFS before and after processing.

Within the DAW signal flow, all faders were kept at unity gain (0.0 dB) level; meanwhile, the signal levels entering the processing chains were equalized using pre-calibrated input gains both in the analog and digital domains (Figure 3).

This approach prevented level discrepancies between signals sent through the analog and software-based processing chains, ensuring a comparable and consistent signal basis for both processing types.

Figure 2

Guitar and Vocals in DAW

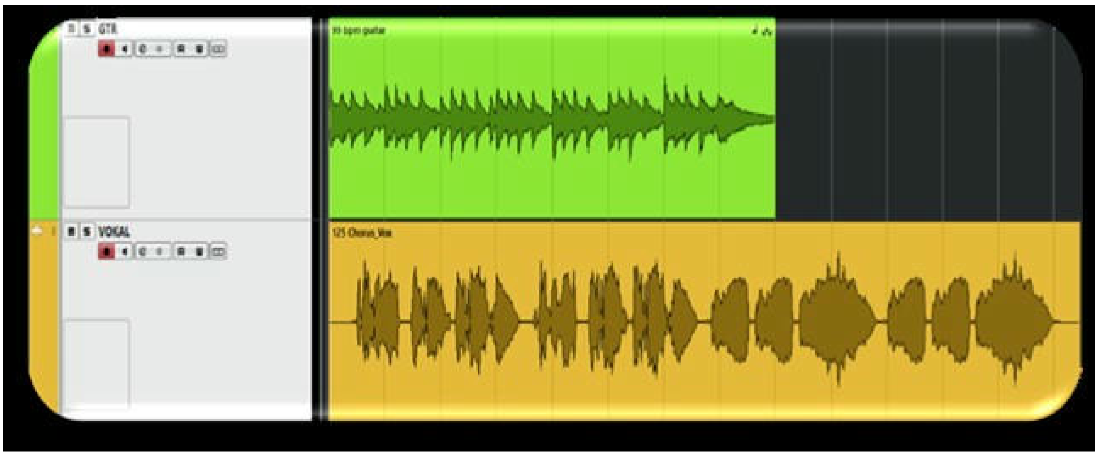


Figure 3

DAW mix settings for Guitar and Vocal

During the processing of audio sources, the audio files used within the DAW were output as analog signals through the Universal Audio Apollo x16 digital converter and then routed back into the designated analog devices via a patchbay. Throughout this process, the audio levels were maintained consistently at 0 dBFS and re-recorded. The gain staging procedure was applied by ensuring that the input signals in both the analog and digital chains remained at nominal levels. Thus, the input gain of the signal was equalized across both processing chains, making the measurements comparable. The audio files were processed through the analog devices in this loop according to the input-output sequence, with the signal level checked at each iteration (Figure 4).

Figure 4

Signal Flow Used

In this study, pre-recorded vocal and guitar tracks were used. These recordings were obtained using high-quality microphones and analog equipment commonly preferred in the music industry. Therefore, they are technically measurable with various analysis software. For this reason, the use of these audio loops in the analyses was deemed appropriate. The vocal and guitar loops were transferred as two separate audio tracks into Steinberg's Cubase 12 digital audio workstation. The first track contains the audio file that was processed through the analog hardware chain and then re-recorded; the second track contains the same audio processed exclusively through software-based digital processors. Both samples were derived from the same audio source under identical sampling conditions, thus ensuring technical equivalence for comparative analysis (Figure 5).

Figure 5

Image in DAW

The prepared loops have durations of 28 seconds for the vocal and 26 seconds for the guitar. Frequency analyses performed on the analog and digital versions of these recordings were conducted over a 30-day testing period using the FabFilter Pro-Q3 plugin. During the analysis, only spectrum visualization was focused on; therefore, no frequency filtering was applied within Pro-Q3. To ensure comparability among all recordings, the RMS levels of both analog and digital samples were equalized using the mvMeter 2 measurement tool prior to analysis, eliminating level-based differences in the spectral results. Additionally, the spectrum images obtained with FabFilter Pro-Q3 correspond to the exact same time frame of the respective tracks, meaning the data were captured simultaneously over the same time segment. This approach prevented inconsistencies that could arise from different time intervals, thereby ensuring the comparability of the spectral analyses. All samples were examined using the Pro-Q3 spectrum analyzer (Figure 6).

Figure 6

Fabfilter Pro-Q3 image

In this study, two different channel loops, vocal and guitar, were each adjusted to a 0 dBFS (decibels Full Scale) reference within the digital audio processing environment. This value represents the highest possible level a signal can reach in the digital domain, and the audio samples were evaluated with the channel faders in Cubase 12 set to unity gain (0.0 dB) position. For signal level and dynamic range measurements, the mvMeter 2 level meter plugin developed by TBProAudio was used, with detailed recordings of RMS and peak values (Figure 7). Both channels, vocal and guitar, were exported as mono audio files in "wav" format at a 44,100 Hz sampling rate and 16-bit resolution. These files were then re-imported into the digital audio processing software for analysis, where both level and frequency values were examined in detail. To ensure fair comparisons, the master output levels of both the analog-processed and digital-processed versions were kept constant at the same 0 dBFS reference, thereby maintaining consistency between noise levels and dynamic measurements.

Figure 7

TB Pro Audio mvMeter-2 image

In the study, the iZotope Insight 2 Pro Metering tool was used for multi-visualizations of stereo field activity and past activity (Figure 8). Within this scope, stereo correlation coefficient measurements between the left and right channels were conducted. Comparative analyses of audio files processed with analog hardware and their digital software simulations were performed using Insight 2 Pro's vectorscope graphs and stereo field measurements. This analysis method provided a more detailed and visual examination of the dynamic structure of the stereo field and the relationships between channels.

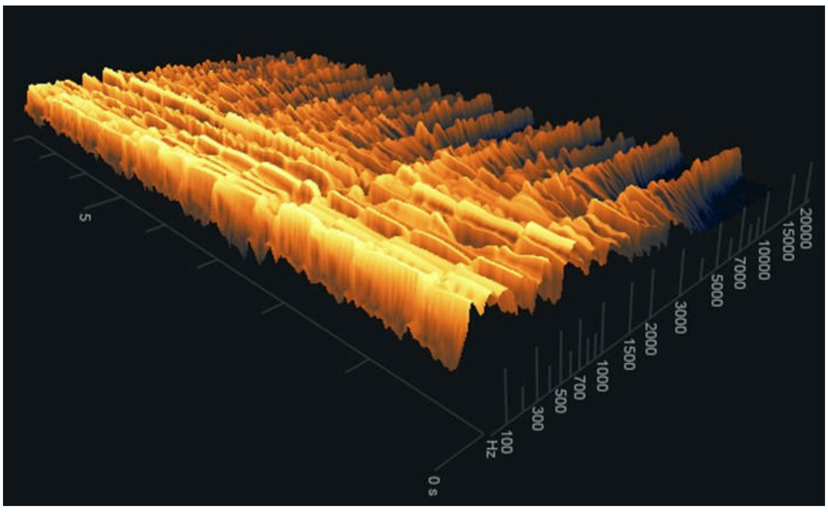
Figure 8

Insight 2 Pro Metering vectorscope image

The iZotope Insight 2 Pro spectrogram tool was used to generate real-time 3D spectrogram graphics, producing detailed topographic sound maps (Figure 9). These maps visually represent the frequency content changes and evolution of signals over time through three-dimensional visualizations, revealing the static, dynamic, and natural timbres of the sounds. Through spectrogram analyses, audio recordings processed by analog hardware equipment and their digital simulations were compared. The purpose of these comparisons was to quantitatively and qualitatively reveal the sound quality differences between analog and digital processing systems and to analyze these differences in frequency-time behavior in detail.

Figure 9

Insight 2 Pro Spectrogram 3D image



In the processing of sound sources, audio files used within the DAW (Digital Audio Workstation) were routed out through a digital converter and then fed back into designated analog devices; during this process, the audio balance was maintained and the signals were re-recorded. The audio files were passed through the analog devices in an "out-input-out-input" sequence. In this study, Steinberg's Cubase 12 digital audio workstation (DAW) was used for the processing of sound sources and the creation of audio analyses. The visual and auditory applications of both analog and digital signal processors were entirely conducted within this software. For both visual and auditory processing, Universal Audio's Apollo x16 A/D and D/A conversion devices were employed, and all audio file processing was performed through this audio interface.

Results

In the findings and comments section, analog-based signal processors were compared with the software-based plug-in versions of these devices. As a result of these comparisons, the fundamental differences between analog hardware processors and software-based plug-ins were detailed.

For the guitar and vocal channels, the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices were used with parameters set to flat and passed through the Neve 8816 Summing mixer. Similarly, UAD DSP and Plugin Alliance software were also used with flat settings and processed through the Neve 8816 Summing mixer (Figure 10).

Figure 10

Insight 2 Pro Spectrogram 3D image

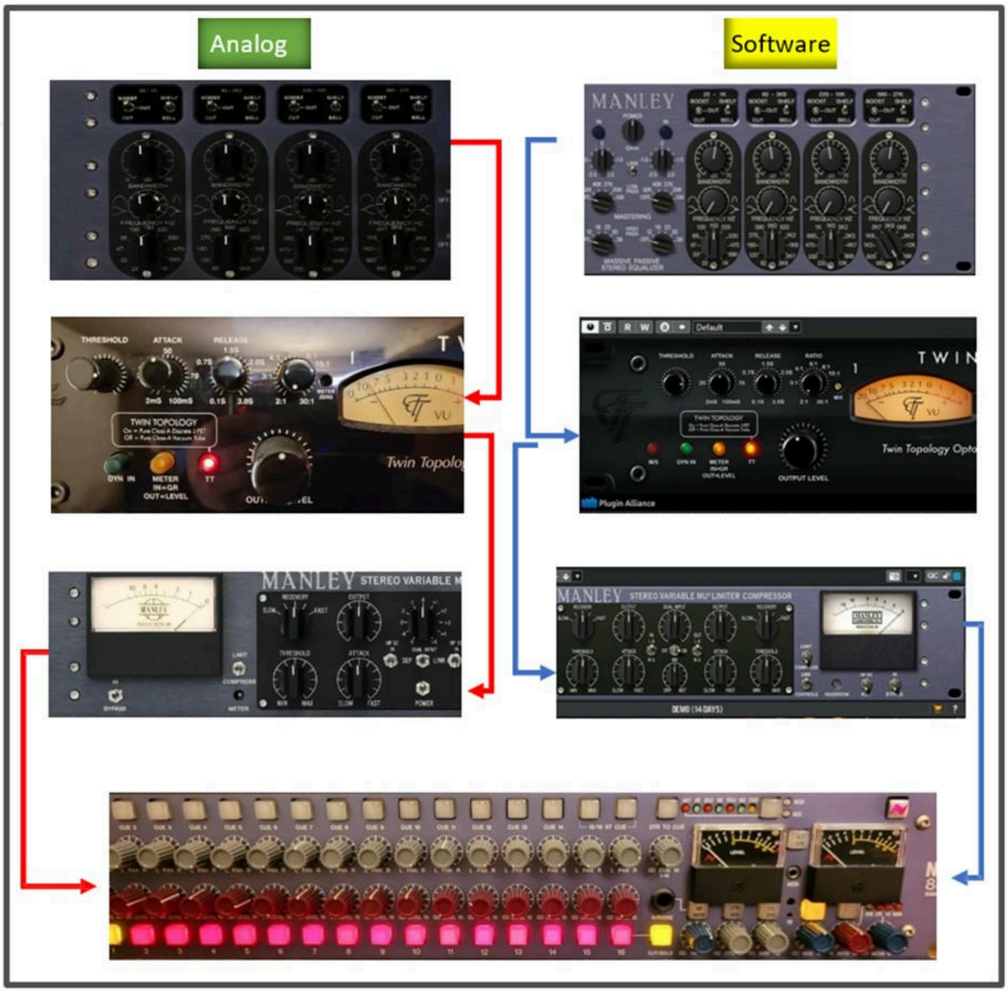
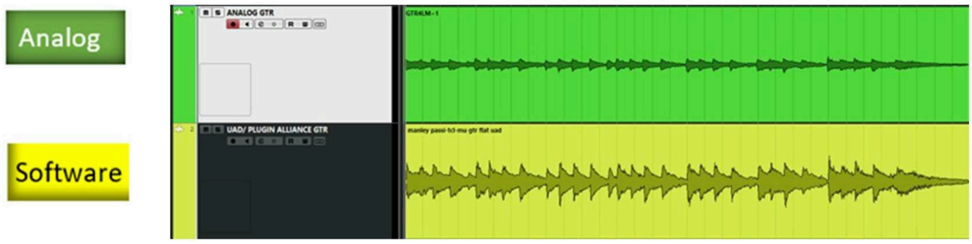


Figure 11

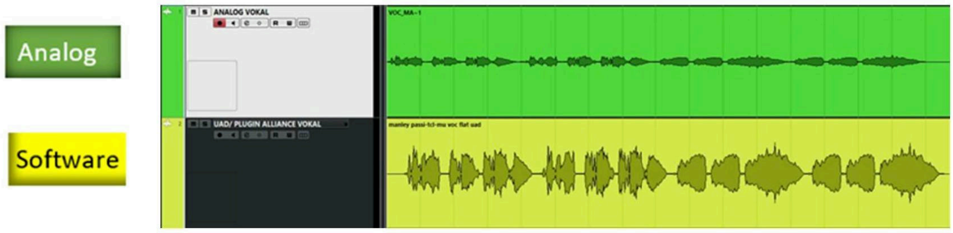
The image of the guitar in the DAW



In the comparison, after obtaining the data from the signal processors, the signal waveforms of the guitar within the DAW are shown in Figure 11. In the first channel, the signal passed through the analog processors is displayed. In the second channel, the signal waveform of the exact software-based processor from UAD and Plugin Alliance is shown. As can be seen from the guitar's display within the DAW, the dynamic range in the second signal is wider compared to the first signal (Figure 11).

Figure 12

Vocal image in DAW



With the same settings, after obtaining the data from the signal processors for the vocal, the signal waveform within the DAW is shown in Figure 12. In the first channel, the signal passed through the analog processors is displayed. In the second channel, the signal waveform of the exact software-based processor from UAD and Plugin Alliance is shown. As can be seen from the vocal's display within the DAW, the dynamic range in the second signal is wider compared to the first signal (Figure 12).

Figure 13

mvMeter-2 maximum peak image of the guitar

After obtaining the data from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices, as well as the software-based signal processor from UAD and Plugin Alliance, the master output fader level in the guitar channel, in dB, and the maximum peak level in mvMeter-2 are shown in

Figure 13. In this visual, the maximum peak level obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices, after compression, was determined to be −9.8 dB using the VU meter measurement. On the other hand, after obtaining data from the analog devices with Universal Audio and the software-based signal processor from Plugin Alliance, the VU meter dB level was measured as −0.2 dB for the maximum peak level. As observed in Figure 202, a sound intensity loss of 9.6 dB was observed between the analog devices and their software-based signal processors (Figure 13).

Figure 14

mvMeter-2 maximum peak image of vocal

After obtaining the data from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices, as well as the software-based signal processors from UAD and Plugin Alliance, in the same way, in the vocal channel, Figure 14 shows that the VU meter of the analog device indicates a level of 2.9 dB, while the maximum peak level shown by the exact software-based signal processors is measured at 10.1 dB (Figure 14). In the vocal channel, a sound intensity loss of 7.2 dB was observed between the analog devices and their software-based signal processors. Differences in dB levels and compression ratios were also observed between the VU meter data obtained from the guitar channel and the vocal channel, between the analog devices and their software counterparts. In the data obtained from the guitar channel, a 9.6 dB sound level difference was observed between the analog and software, while in the vocal channel, a 7.2 dB sound difference was observed. These differences highlight the diversity in sound characteristics and performances between analog and software-based signal processing tools.

Figure 15

Spectrum analysis image of the guitar (Analog)

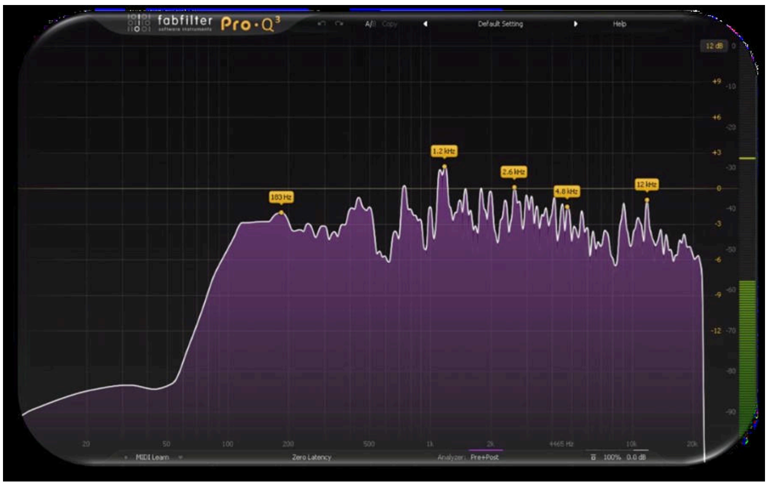


Figure 16

Spectrum analysis image of the guitar (Software)

Figure 15 shows the spectrum analysis of the guitar channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices (Figure 15). According to this analysis, it was observed that the guitar has frequency components in the range of approximately 100 Hz to 20 kHz, with the most prominent frequency regions concentrated around 183 Hz, 1.2 kHz, 2.6 kHz, 4.8 kHz, and 12 kHz. The analysis reveals that analog devices such as the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 control the dynamic range of the signal without cutting off the low-frequency range of the guitar.

Figure 16 shows the spectrum analysis of the guitar channel data obtained from the UAD and Plugin Alliance software (Figure 16). According to this analysis, it was observed that the guitar has frequency components in the range of approximately 100 Hz to 20 kHz, with the most prominent frequency regions concentrated around 172 Hz, 1.2 kHz, 2.6 kHz, 4.8 kHz, and 12 kHz. However, some frequency values differ between the analog devices and the software. These differences reflect the varying frequency interventions and sound characteristics between the analog devices and the software. While the software can make certain frequencies more prominent, the analog devices provide a broader dynamic control.

Figure 17

Spectrum analysis image of vocal (Analog)

Figure 18

Spectrum analysis image of vocal (Software)

Figure 17 shows the spectrum analysis of the vocal channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices (Figure 17). This analysis shows that the vocal has frequency components in the range of approximately 100 Hz to 20 kHz. The most prominent frequency regions in the vocal channel were observed to be around 420 Hz, 820 Hz, 1.7 kHz, 3.7 kHz, and 6.9 kHz. It was determined that analog devices such as the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 control the dynamic range of the signal without cutting off the low-frequency range of the vocal. This means that analog devices cause less interruption in frequency interventions while preserving the natural timbre of the vocal. Figure 18 shows the spectrum analysis of the vocal channel data obtained from UAD and Plugin Alliance software (Figure 18). This analysis reveals that the vocal has frequency components in the range of approximately 200 Hz to 20 kHz, with the most prominent frequency regions concentrated around 432 Hz, 820 Hz, 1.7 kHz, 3.8 kHz, and 6.9 kHz. However, compared to the analog devices, some frequency differences were observed in the data obtained from the software, with a cutoff applied between 0 and 70 Hz. This shows that software-based processors tend to provide more control by cutting specific frequencies, and by limiting the low frequencies, they make the signal cleaner and more controlled.

Figure 19

Vectorscope image of the guitar

Figure 19 shows the vectorscope graph area measurements of the guitar channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices, as well as the software-based versions of the same analog devices (Figure 19). This graph illustrates the maximum stereo field activity of the guitar and shows the stereo correlation measurements between the left and right channels. The analysis revealed that the maximum stereo field activity is distributed in a 360-degree surround format.

However, in the comparison made through the graph, it was observed that the guitar sound did not spread across the same wide area with the software-based signal processor as it did with the analog devices. This suggests that the analog devices provide a more natural and expansive stereo field, while the software-based processors may limit the stereo field activity to a narrower range or present it with a different character. This difference could have created varied tonal and width effects, especially in the stereo image and the placement of the sound.

Figure 20

Vectorscope image of vocal

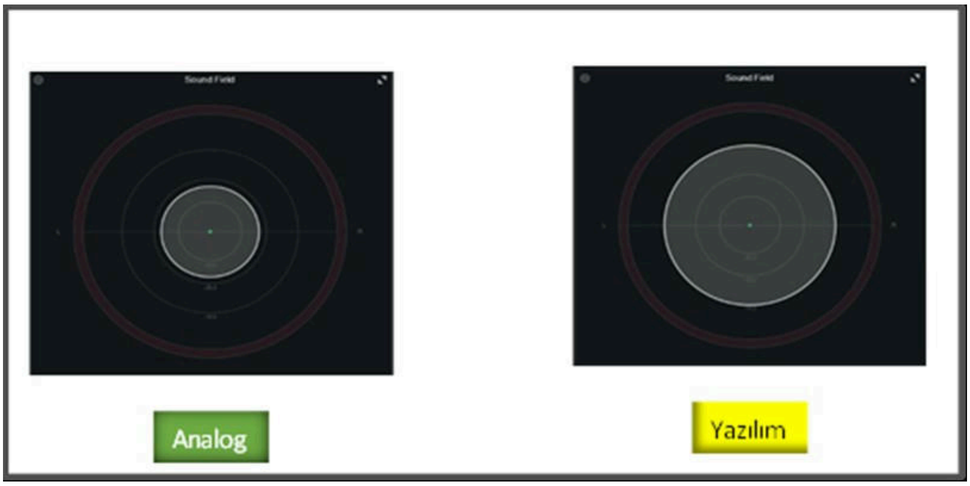


Figure 20 shows the vectorscope comparison graph of the vocal channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices, as well as the software-based versions of the same analog devices (Figure 20). This graph illustrates the maximum stereo field activity of the vocal in a 360-degree surround format.

As a result of the comparison, it was observed that the vocal sound did not spread across the same wide stereo field with the software-based signal processor as it did with the analog devices. The analog devices provide a wider and more natural stereo field, while the software-based processors exhibit a more limited spread in this regard. This difference may have impacted the width and placement of the vocal's stereo image, potentially creating a significant difference in terms of the vocal's positioning within the mix and the listening experience presented to the audience.

Figure 21

Spectrogram graph of the guitar (Analog)

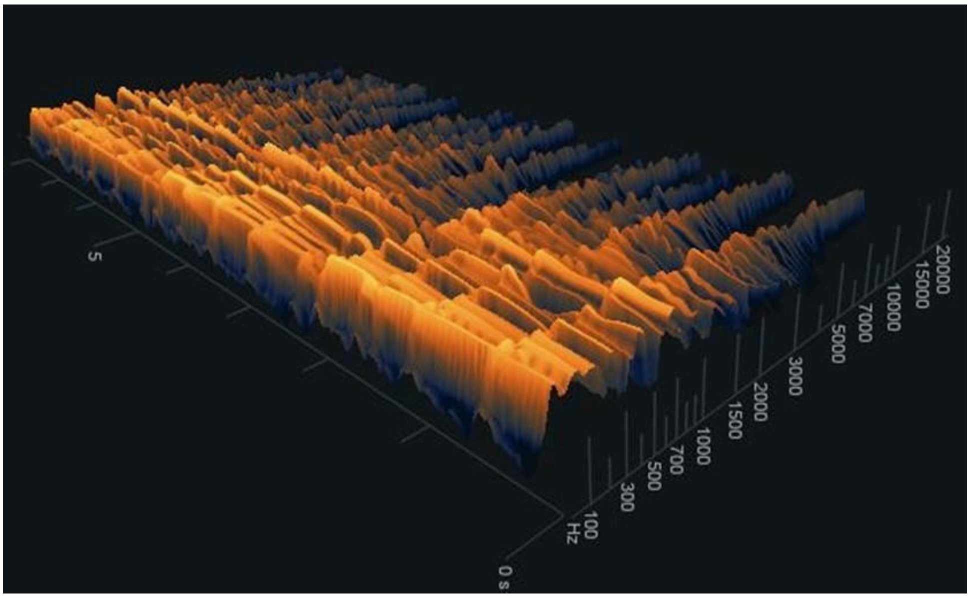


Figure 22

Spectrogram graph of the guitar (Software)

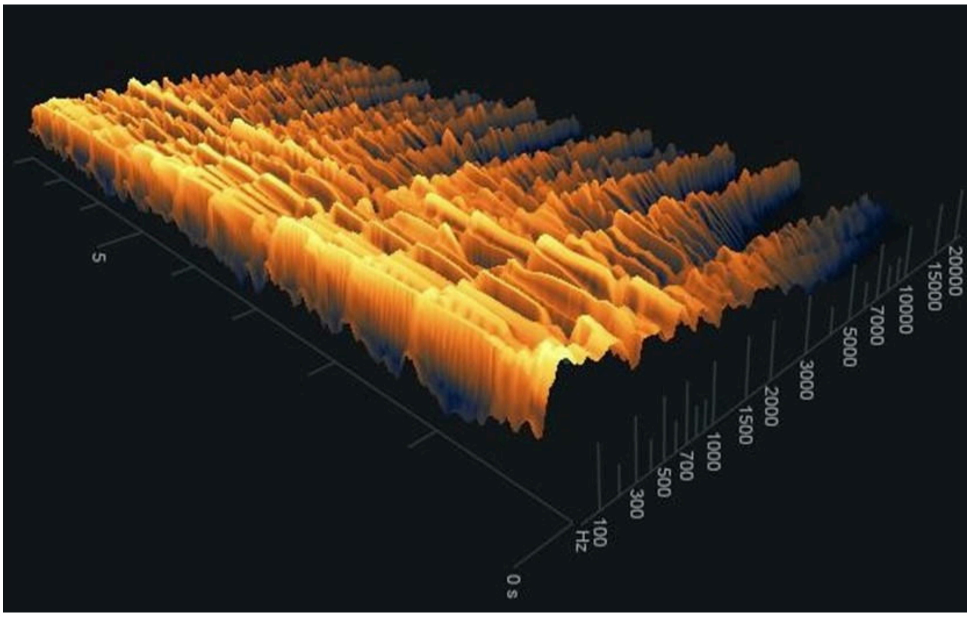


Figure 21 displays a 3D spectrogram of the guitar channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices (Figure 21). This graph shows the detailed topographic sound map of the guitar. The signal's static, dynamic, and natural timbre in the data from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 devices is displayed according to the expected graphical shape over time. The analog devices accurately reflect the guitar's natural sound characteristics, providing a balanced performance across the dynamic range.

Figure 22, on the other hand, shows the 3D spectrogram of the guitar channel data obtained from UAD and Plugin Alliance software-based signal processors (Figure 22). Compared to the data from the analog devices, this graph indicates that energy levels between 100 Hz and 3000 Hz are higher. Software-based processing generally spreads across a broader frequency range, potentially displaying a more energetic character than the natural timbres offered by the analog devices.

The spectrogram analysis highlighted the differences between analog devices and software-based processors. The analog devices in Figure 21 focus on lower frequencies (100 Hz to 3000 Hz) with less energy, while the software-based models Figure 22 exhibit higher energy in the same frequency range, creating a more prominent sound saturation (Figure 22). This difference provides important insights into how the processors handle the audio signal, in addition to revealing the dynamic and static characteristics of the sounds.

Figure 23

Spectrogram graph of vocal (Analog)

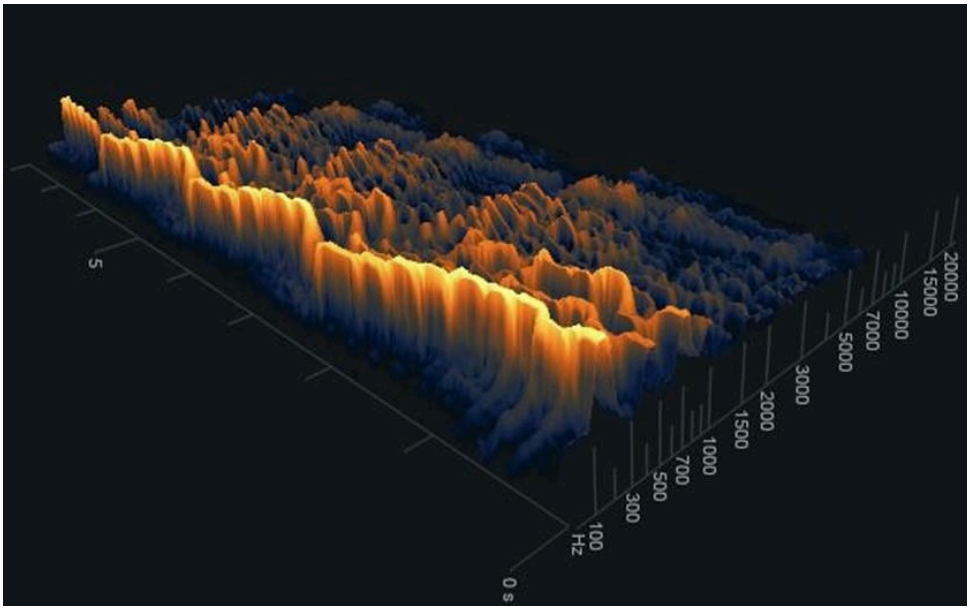


Figure 24

Spectrogram graph of vocal (Software)

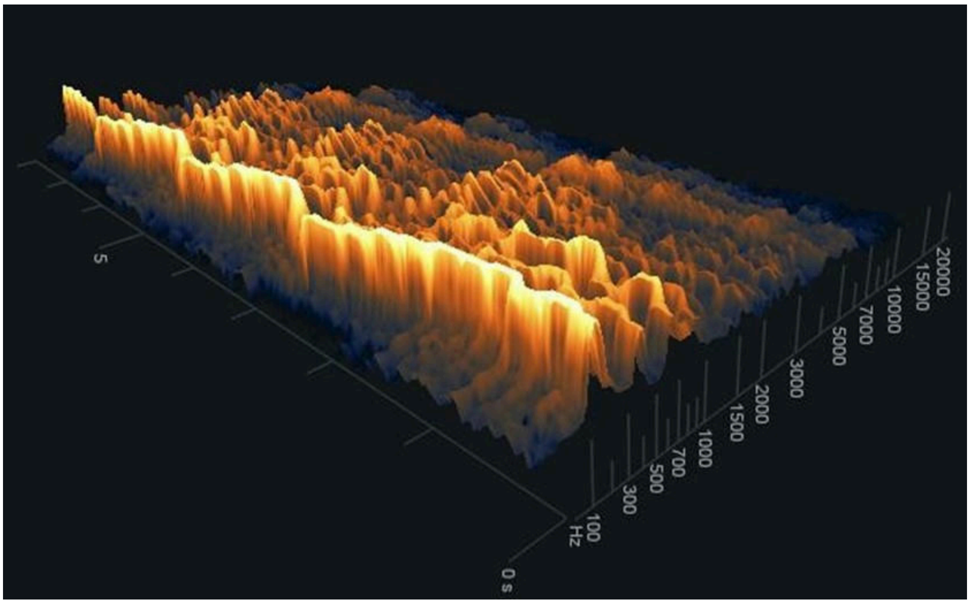


Figure 23 displays a 3D spectrogram of the vocal channel data obtained from the Manley Passive, Manley Variable Mu, and Millennia Music TCL-2 analog devices (Figure 23). This graph presents the detailed topographic sound map of the vocal, visualizing how the signal's static, dynamic, and natural timbre changes and progresses over time. The analysis shows how the analog devices process the audio signal, the changing frequency components over time, and the overall nature of the signal. This graph reflects how analog processors control the signal and how the sound naturally evolves over time.

Figure 24 shows the 3D spectrogram of the vocal channel data obtained from UAD and Plugin Alliance software-based processors (Figure 24). This graph reveals how the software-based processors handle the signal, particularly how they differentiate the energy levels across frequency ranges. In the spectrogram from the analog devices in Figure 23, the dynamic range of the signal is focused on lower energy levels between 100 Hz and 1000 Hz, while in the software-based processors (Figure 24), higher energy levels are observed in the same frequency range.

This difference highlights the fundamental characteristic differences between analog devices and software. The analog devices in Figure 23 focus on less energy in the lower frequencies and preserve the natural timbre more, whereas the software-based processors present a more intense sound with higher energy levels in the same frequency band. Additionally, it was observed that the saturation values of both analog devices and software-based processors differ in both visuals, further emphasizing how distinct the sound production and processing methods are between these two technologies.

Discussion, Conclusion, and Recommendations

In this study, the effects of analog and software-based signal processors on the mixing stages were compared. The findings reveal significant differences between the two types of processors. Notably, analog processors provide a natural harmonic coherence that adds a richer and warmer tone to the signal, whereas this coherence could not be fully achieved in software-based processors. Guitar and vocal signals processed through analog devices exhibited higher peak levels and dynamic range at the master output, while these levels were noticeably lower in software-based processors. Additionally, spectrum analyses and frequency component evaluations showed that analog devices offer a wider frequency range, whereas certain frequency bands suffered energy loss in software-based processors. This indicates that analog and software-based signal processors differ fundamentally in frequency energy distribution and dynamic behavior.

Furthermore, the saturation levels of analog processors varied throughout operation, whereas those of software-based processors remained constant. This demonstrates that the warmth and character provided by analog devices cannot be fully replicated in software-based processors. Spectrogram analyses revealed that signal energy was concentrated between 100 Hz and 2000 Hz in analog processors, but energy loss was observed in this frequency range in software-based processors. These results highlight the core differences between analog and digital technologies and illustrate how the sonic characteristics they deliver are distinctly different.

Supporting these qualitative observations of "warmth," harmonic distortion measurements and harmonic content analyses were conducted, confirming that analog processors introduce more desirable harmonic distortions, contributing to their perceived warmth and richer tone. These quantitative analyses provide objective backing to the subjective listening impressions reported in this study.

Based on the findings of this research, the following recommendations are made for various music production environments and requirements:

Selection According to Usage Purpose

For live recordings and projects requiring rich harmonic structures, analog signal processors are recommended. Analog devices add a more natural and warm character to the signal, enhancing harmonic distortions that result in a more satisfying tone.

For electronic music genres, especially dance, techno, and drum 'n' bass, where a cleaner and lower-distortion sound is preferred, software-based signal processors may be used. These processors offer a cleaner and more neutral signal suitable for such production needs.

Cost and Physical Space Constraints

Analog processors, being physical units, can be costly and pose challenges in portability and studio setup due to their size. These devices often require significant physical space and can strain the budget.

Software-based processors are more economical compared to analog units and run on computer systems, making them portable and easily accessible. Therefore, they provide a suitable alternative for studios with limited budgets and space constraints.

Technological Advances and Updates

Software-based signal processors are continuously updated and developed by manufacturers, enabling them to offer new features and improvements over time. Given the rapidly evolving nature of technology, current software-based processors provide audio engineers with new opportunities and capabilities.

Analog processors rely on physical components, which require hardware modifications for upgrades. Consequently, audio engineers who want to keep up with technological changes and access new features may find software-based processors more adaptable.

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