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Research Article

Comparison of analog processors and digital signal processors

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Abstract

Mastering, on the other hand, is the process of refining a recording made after the mixing stage using various techniques before the album is pressed and distributed. The mastering process includes adjusting the dynamics of signal frequencies, regulating tonal balances with equalizers, and determining and downsizing audio file formats. Mastering creates coherence among the tracks within an album and provides listeners with a higher-quality listening experience. The entire process of recording and mixing, encompassing mastering, involves converting signals from analog equipment and software into digital values using an Analog-to-Digital (A/D) converter within a Digital Audio Workstation (DAW). These software programs have become essential in music productions. Due to the decreasing cost of technology and the opportunities it provides, music production software has shifted towards home users. People can now, without the need for high-budget studios, complete many recording, mixing, and mastering processes entirely with computer-based systems at lower costs in their own homes. The term "in the box" refers to all these production stages taking place within a computer. With the advancement of technology, individuals have been able to produce albums in home studios, and digital processors, which are cheaper and more practical than analog equipment due to the digitization of analog devices, have started to be preferred over analog equipment. This study explores the extent of changes in technical and technological approaches towards the use of digital signal processors (DSP), which emerged in the 1960s, in place of analog processors. The research aims to identify and examine the differences between digital-based signal processors and analog devices in terms of their usage, implementation, and processing in music technologies. This research investigates how digital-based signal processors differ from analog processors in terms of processing method, processing flexibility, processing speed, processing quality, and sensitivity on the audio signal. Based on these findings, it is concluded that digital-based signal processors, which are replacing analog processors in the music industry, do not possess the same qualities as analog devices.

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Introduction

The history of sound recording technology dates back to the early 1800s and began to develop in the mid-19th century as a result of technological advancements. In 1807, American Thomas Young created a device capable of transmitting acoustic vibrations onto a cylinder, and in 1857, Leon Scott de Martinville invented a device called the phonautograph, which had the ability to record sounds. Using an old method, he darkened paper with the heat of a gas lamp to record the sound waves, successfully capturing a French folk song on April 9, 1860 (Cumhuriyet Bilim Teknik, 1994). The

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phonograph, invented by Thomas Edison in 1877, was referred to as a "talking machine" and allowed for the recording and playback of sound. Edison's phonograph marked the first visual representation of sound in recorded history (CNNTÜRK, 2008). Following this invention, Alexander Graham Bell in 1886 and Emile Berliner in 1888 obtained patents for the phonograph (Mc Queary, 1990). The first experiments with magnetic tape recording were conducted in 1888 by Oberlein Smith, and the first magnetic tape recording in 1889 by Danish physicist Valdeman Poulsen using the "telegraphone" (Mumma, Rye, and Kernfeld, 2010). The microphones of telephones used for communication by people contained carbon materials, resulting in uneven frequencies of sounds and signals. In music productions, radios, and recordings, the microphones of telephones caused excessive background noise and were not used in radio broadcasts and music recording studios. The Western Electric Engineering Department developed a dual-button carbon microphone, reducing the background noise in microphones. As a result, these microphones began to be used in radio broadcasting and music productions (Burgess, 2014). Mechanical recordings made during that period experienced acoustic power loss, leading to a decrease in the frequency response of sound and an increase in the level of background noise on the recording. This adversely affected the quality of sound. In the early 20th century, engineers at Bell Laboratories and contemporary scientists began working on electronic sound recording to improve the quality of sound recordings. With the contributions of the Western Electric Engineering Department, an electromechanical recording device was designed, replacing the recording needle with a condenser microphone. In 1925, the first company to obtain a license for electronic recording was the Victor Talking Machine Company (Schmidt-Horning, 2013). BASF and TDK companies produced the first magnetic oxide-coated tapes in 1930. In 1935, AEG introduced the first magnetic tape recording device called "Magnetophone." With rapidly advancing technology, the quality of recorded sounds improved, making sound recordings more practical. The development of vinyl technology led to the emergence and popularity of 45 and 33 1/3 RPM records in the early 1960s. This significantly contributed to the commercial progress of the music industry. In 1955, Les Paul collaborated with Ampex to create an 8-channel recording device. Ampex manufactured the device in 1956, and Les Paul laid the foundation for the channel recording techniques used in today's recording world. Therefore, 1956 is recognized as the year when the first channel recording device was created (Önen, 2010).

Compressor

A compressor is a device that enables automatic control of sound levels (Bartlett, 2005). It applies compression to signals that exceed a predetermined signal threshold, reducing level differences in signals and balancing the output level (Durmaz, 2009). Compressors equipped with a light source are referred to as optical (opto) compressors. As the sound level increases, the brightness of the light source inside the compressor also increases. Conversely, as the signal's sound level decreases, the brightness of the light diminishes (Coşaner, 2008). The initial use of compressors began in the 1930s with the Western Electric 110A series in the United States and became widespread worldwide (Pasinlioğlu, 2016). On the other hand, a limiter reduces the signal as a limiting device and prevents it from exceeding the peak level (Edstrom, 2011). Additionally, a limiter prevents unnecessary signals from distorting sounds during recording (Öcek, 2010). An expander raises the lower levels of the signal, expanding the dynamic range and making lower sounds more pronounced. In short, it aims to emphasize the portions of the signal that fall below a certain level, thereby increasing the dynamic range (Izhaki, 2008).

In today's audio systems, sampling frequencies of 44,100 Hz and above are used as the upper hearing limit for a healthy individual is 20,000 Hz (Rumsey 1994). The focal point of music production extends beyond merely performing or creating a musical piece; it encompasses the recording and processing of the recorded material (Lefford, 2000). Throughout this process, the audio recording industry has undergone a fundamental transformation due to advancing technology. Despite debates between analog and digital technologies, digital technology has become dominant, leading to a transformation in the music production process (Huber, 2010). A debate has emerged regarding the comparison between analog equipment and digital processors, which offer convenience and cost-effectiveness.

Problem of Study

It is important in terms of revealing the level of effectiveness of the techniques applied by using various tools in the production process. This study aims to reveal the extent of differences in technical and technological approaches between analog processors used in the recording, mixing, and mastering processes and their digital counterparts. By focusing on the use of signal processors in digital music, this research aims to identify the similarities and differences of the software offered by today's computer technology compared to analog devices. The problem of study is;

- What are the differences between analog processors and software-based processors in the production process?

Method

Research Model

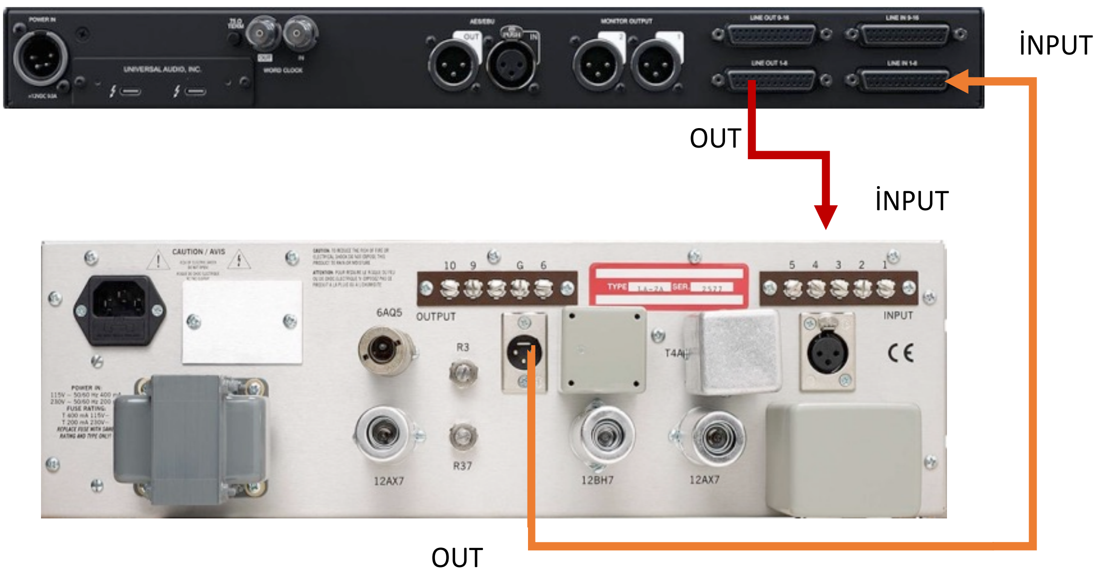
This research has been conducted using the qualitative research method of literature review, and the obtained results have been evaluated and presented in line with the problem of the study. In this study, which aims to highlight the differences between software used in computer technology and analog devices, a comparison based on the principle of matching data with auditory and visual materials has been made.

The qualitative research method involves the use of information collection methods such as observation, interviews, and document analysis. This method aims to present perceptions and events in a natural environment in a realistic and holistic manner (Yıldırım and Şimşek, 2005).

Data Collection Tool

In the research, the Cubase 12 digital audio workstation (DAW) by Steinberg has been utilized for processing audio sources and creating sound analyses. The entire visual and auditory applications essential to the core of the study, comparing analog signal processors with digital signal processors, have been executed within this software. For the visual and auditory applications of sound files, the Apollo x16 A/D and D/A conversion by Universal Audio has been employed.

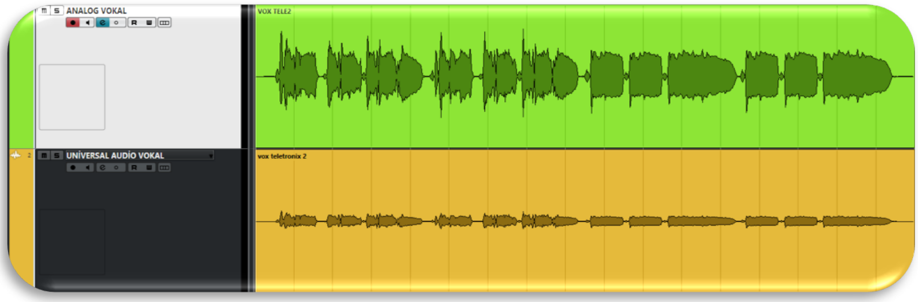
In the processing of audio sources, the digital files used within the DAW were output through a digital converter, and when re-entered into the input section of the designated analog devices, the sound balance was consistently re-recorded. The sound files were created in an iterative process within this loop of output-input-output-input sequence through analog devices.



Picture 1. The visualization of the signal within Cubase DAW

Tools and Materials

In the research, the processing of audio sources was conducted using the Teletronix LA-2A analog processor, specifically the Leveling Amplifier, and its one-to-one simulation, the UAD Teletronix LA-2A Classic signal processor. Pre-prepared vocal and guitar tracks were selected and processed in mono (single-channel) format. The parameters of the audio sources were kept constant at 0 dB.



Picture 2. The visualization of the signal within Cubase DAW

Picture 3. Guitar and vocal mixing settings

Preparation of the Project File

In the research, two previously prepared different channel loops, vocal, and guitar, were selected and created in mono (single-channel) format. Vocal and guitar loops are among the most commonly used channels in the industry. These channels were selected because they can be measured with various analysis programs as they are recorded with microphones and analog devices. They were transferred into Steinberg's Cubase 12 digital audio workstation as two audio tracks. The first channel had the audio file that had previously passed through the analog equipment. The second channel had the version of the same audio that had previously passed through the software. The prepared loops for vocal and guitar are 28 seconds and 26 seconds long, respectively. Frequency analyses were performed on the recordings created from analog and the same analog recordings processed through software. The analyses were conducted using the Fabfilter Pro-Q3 equalizer spectrum analyzer, an industrial-grade tool, during a 30-day trial period. In this process, no filtering was applied to the equalizer; only the spectrum analysis images were used. The Pro-Q3 spectrum frequency analyzer was used for all audio recording files created from analog and the same analog sources.

Picture 4. Screenshot of the FabFilter Pro-Q3 plugin

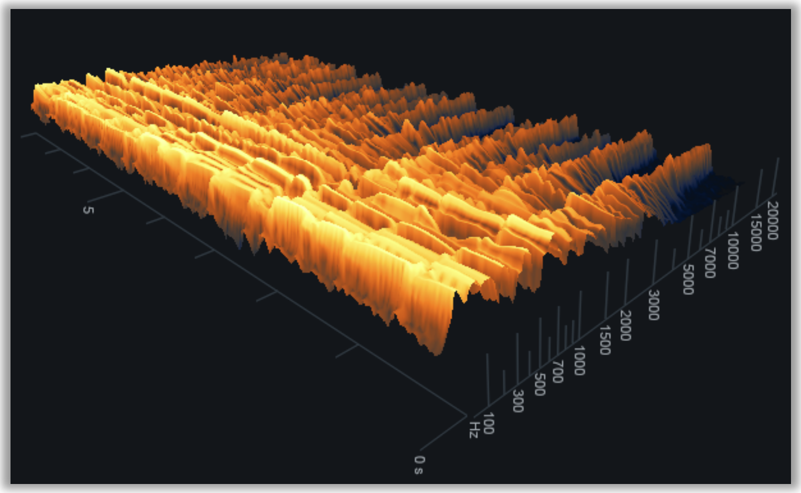
In the research, the dB level of the two different channel loops, vocal and guitar, was set to 0 (zero). To measure the loudness and dynamic values, a free plugin called "mvMeter 2" from TB Pro Audio was used. The sample rate for both prepared channels, vocal and guitar, was set to 44,100 samples per second, and the recordings were saved in the "wav" mono audio file format with a 16-bit resolution. The audio was exported from the recording station to the outside. The exported audio files were then re-imported into the digital audio workstation (DAW), and the audio and frequency values were examined. The master output of the channels was fixed at 0 dB, and the loudness levels were calculated equally for both analog and software processes.

Picture 5. Screenshot of the TB Pro Audio mvMeter-2 plugin

In the research, Insight 2 Pro Metering from Izotope was used for multiple visualizations of stereo field activity and history. The stereo correlation measurements between the left and right channels were evaluated. Processed audio files from both analog and the corresponding software were analyzed using the vector scope measurements in the Insight 2 Pro Metering tool..

Picture 6. Screenshot of the Insight 2 Pro Metering vector scope plugin

Real-time 3D spectrogram graphics were created using Insight 2 Pro Spectrogram from iZotope, generating detailed topographic sound maps. In these sound maps, 3D visual graphics were used to measure the dynamic and static states of sounds, showing how the frequency content of signals changes over time. The progression of signals was observed in 3D visual graphs. Comparisons were made between the spectrogram graphics of analog and the corresponding analog equipment in all audio files. Through these analyses, differences between analog hardware equipment and their one-to-one software simulations were identified.



Picture 7. Screenshot of the Insight 2 Pro Spectrogram 3D plugin

The audio sources were recorded by taking the output through the digital converter within the Digital Audio Workstation (DAW), sending it out from the DAW, passing it through the designated analog devices, and then re-recording it by inputting it back into the DAW. In the processing of audio files, this loop was created within analog devices following the sequence of out-input-out-input. For the processing of audio sources and the creation of audio analyses in the research, Steinberg's Cubase 12 digital audio workstation was used. All visual and auditory applications of analog and digital signal processors, which form the essence of the study, were carried out within this software. For the visual and auditory applications of audio files, Apollo x16 A/D and D/A conversion from Universal Audio were utilized. All visual and auditory applications of audio files were also performed using this sound card.

Sampling

The universe of the research comprises all software programs, including music software, and the computers running these programs. The sample consists of Digital Audio Workstation (DAW) programs, which encompass digital audio processing stations.

Findings

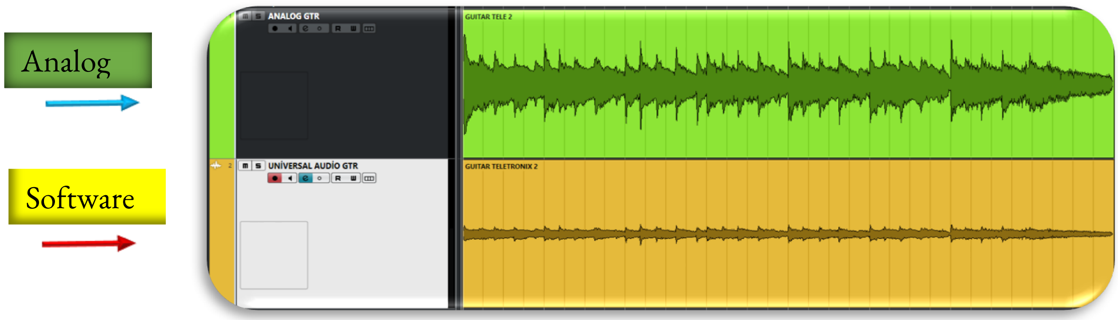
Comparison of Teletronix LA-2A and Universal Audio DSP

Picture 8. Teletronix LA-2A parameters

For the guitar and vocal channels, the compressor parameters of the Universal Audio Leveling Amplifier, Teletronix LA-2A analog device, have been set as follows: Gain: 40, Peak Reduction: 70.

Picture 9. UAD Teletronix LA-2A parameters

The UAD Teletronix LA-2A Classic Leveling Amplifier, which is a direct simulation of the Universal Audio Teletronix LA-2A model, has been configured with the same settings for both the guitar and vocal channels, specifically Gain: 40 and Peak Reduction: 70.



Picture 10. The view of the guitar within Cubase DAW

Following the comparison of the data obtained from the signal processors, the signal forms of the guitar within the DAW are presented in Figure 8. The first channel displays the signal that passed through the analog processor. In the second channel, there is the signal representation of the one-to-one simulation from the UAD company. As evident from the visual representation of the guitar signal within the DAW, it can be observed that, compared to the first signal, the dB value has decreased in the second signal, indicating a weakening in the dynamic range of the signal.

Picture 11. The view of the vocal within Cubase DAW

With the same settings, the signal representation within the DAW after obtaining data from the vocal signal processors is shown in Figure 9. The first channel displays the signal that passed through the analog processor. In the second channel, there is the signal representation of the one-to-one simulation from the UAD company. As evident from the visual representation of the vocal signal within the DAW, it can be observed that, compared to the first signal, the dB value has decreased in the second signal, indicating a weakening in the dynamic range of the signal.

Picture 12. The maximum peak view of the guitar's mvMeter-2 plugin

After obtaining data from the Teletronix LA-2A analog device by Universal Audio and its UAD Teletronix LA-2A Classic simulation, the master output fader level of the guitar channel was measured by assessing the maximum peak level using mvMeter-2, as shown in Figure 10. In this visual representation, the maximum peak level of the data obtained from the Teletronix LA-2A analog device was measured as 6.8 dB using the VU meter. On the other hand, the VU meter dB level of the data obtained from the UAD Teletronix LA-2A Classic simulation of the analog device was measured at a maximum peak level of -9.9 dB. As evident from Figure 10, there is a 16.7 dB loss in sound intensity between the analog device and its simulation.

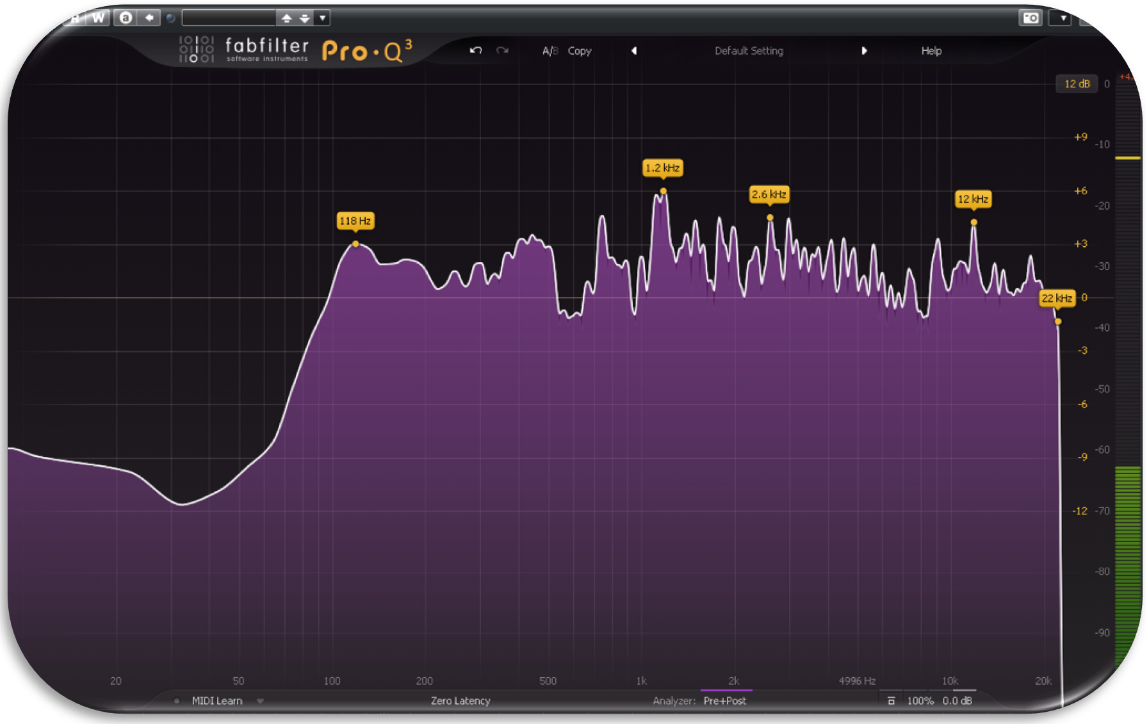
Picture 13. The maximum peak view of the vocal's mvMeter-2 plugin

The data obtained from the vocal channel using the Universal Audio Leveling Amplifier, Teletronix LA-2A analog device, and its UAD Teletronix LA-2A Classic simulation are illustrated in Visual 11. In this illustration, while the VU meter of the analog device indicates a level of 8.4 dB, the maximum peak level shown by the simulation of the device is measured at -6.1 dB. In the vocal channel, there is a 14.4 dB loss in sound intensity between the analog device and its simulation.

Discrepancies in dB ratios between the VU meter data obtained from the guitar channel and the data obtained from the vocal channel and the software simulation of the same analog equipment have been observed. While there is a 16.7 dB difference in sound levels between analog and software in the data obtained from the guitar channel, there is a 14.4 dB difference in the data obtained from the vocal channel.

Visual 12 displays the spectrum analysis of the data obtained from the guitar channel of the Teletronix LA-2A analog device. According to this analysis, the guitar has frequency components in the range of approximately 100 Hz to 20 kHz. The most prominent frequency regions of the instrument, as observed from the spectrum analysis, are concentrated around 118 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 cutting into the lower-frequency range of the guitar.

Visual 13 shows the spectrum analysis of the data obtained from the guitar channel of the Teletronix LA-2A software, which is a one-to-one simulation by UAD. Similar to the data from the analog device, the guitar exhibits frequency components in the range of approximately 100 Hz to 20 kHz. According to the spectrum analysis, the most prominent frequency regions of the instrument are concentrated around 183 Hz, 1.2 kHz, 2.6 kHz, 4.8 kHz, and 12 kHz, as observed in the analog data. Compared to the analog device, a loss in intensity is observed in all frequency values with the data obtained from the software simulation of the same analog device. Additionally, the software simulation has applied an automatic low-cut process to the guitar channel for frequencies below 20 Hz to 50 Hz.



Picture 14. Spectrum analysis of the guitar with the Fabfilter Pro-Q3 plugin (Analog)

Picture 15. Spectrum analysis of the guitar with the Fabfilter Pro-Q3 plugin (Software)

In Visual 14, the spectrum analysis of the data obtained from the vocal channel of the Teletronix LA-2A analog device is presented. 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 18 kHz. The Teletronix LA-2A analog device is measured in the spectrum analysis, showing control over the dynamic range of the signal without cutting into the lower-frequency range (20 Hz to 200 Hz) of the vocal.

Visual 15 displays the spectrum analysis of the data obtained from the vocal channel of the Teletronix LA-2A software, which is a one-to-one simulation by UAD. Similar to the data from the analog device, the vocal exhibits frequency components in the range of approximately 100 Hz to 20 kHz. According to the spectrum analysis, the most prominent frequency regions of the vocal are concentrated around 420 Hz, 1.7 kHz, 3 kHz, 10 kHz, and 18 kHz, similar to the analog data. The data obtained from the software simulation shows a loss in intensity across all frequency values in the spectrum analysis. Additionally, it is evident from the software simulation that a low-cut process has been applied to the vocal channel for frequencies approximately between 20 Hz and 100 Hz.

While there is no cutting or frequency loss in the lower frequencies in the analog device, the impact created by the software simulation of the same analog device is observed to be different and inconsistent for both the guitar and vocal channels. In the data obtained from the software, the cutting frequency range for the guitar channel is between 20 Hz and 50 Hz, while for the vocal channel, the cutting frequency range is observed to be between 20 Hz and 100 Hz. The impact applied by the Teletronix LA-2A software simulation by UAD on both channels is observed to be different and not consistent.

Picture 16. Spectrum analysis of the vocal with the Fabfilter Pro-Q3 plugin (Analog)

Picture 17. Spectrum analysis of the vocal with the Fabfilter Pro-Q3 plugin (Software)

Picture 18. Vector scope view of the guitar with the Insight 2 Pro Metering plugin

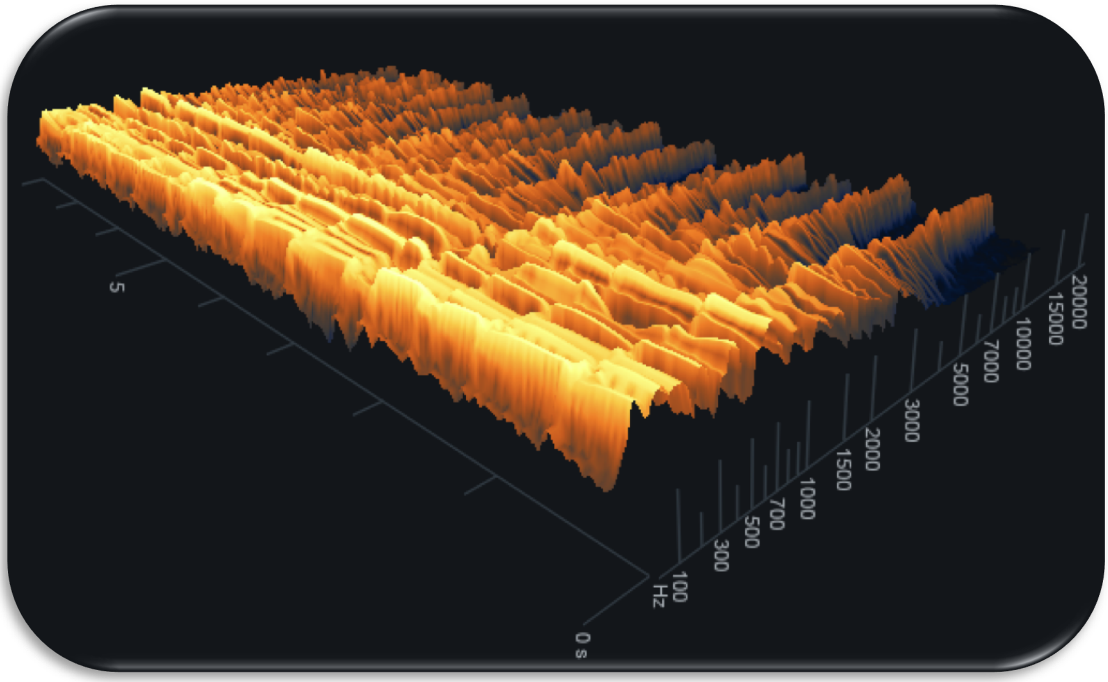
In Visual 16, the vector scope graph measurements of the data obtained from the guitar channel of the Teletronix LA-2A analog device and the software simulation are shown. The vector scope graph depicts the maximum stereo field effectiveness, and measurements of stereo correlation between the left and right channels have been conducted. By displaying the maximum stereo field effectiveness in a 360-degree surround distribution, it is observed that, compared to the software simulation, the analog device spreads the guitar sound over a broader area.

Picture 19. Vector scope view of the vocal with the Insight 2 Pro Metering plugin

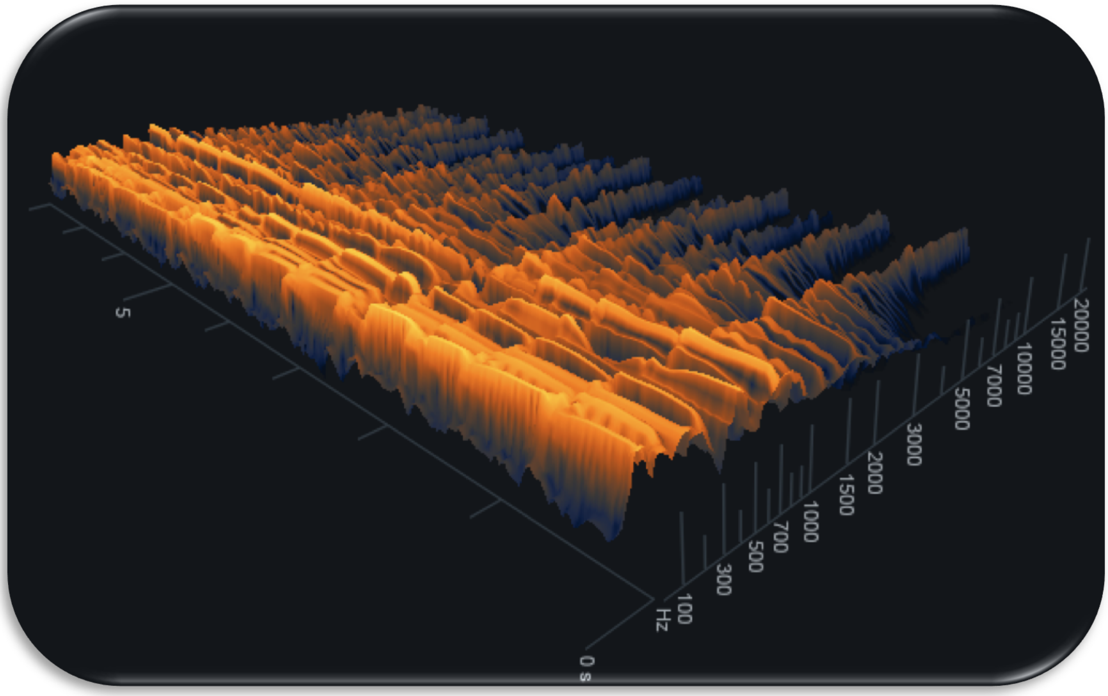
In Visual 17, the vector scope comparison graph of the data obtained from the vocal channel of the Teletronix LA-2A analog device and the software simulation is displayed. In the 360-degree surround distribution, representing the maximum stereo field effectiveness, it is observed that the analog device spreads the vocal sound over a wider area compared to the software simulation.

The three-dimensional spectrogram graph of the data obtained from the guitar channel of the Teletronix LA-2A analog device is shown in Image 18. In this analysis, a detailed topographic sound map of the guitar is created. In the spectrogram graph of the guitar instrument in Image 18, after obtaining data from the Teletronix LA-2A analog device, the graph shape that the signal's static, dynamic, and natural tone should have over time is displayed.

In Image 19, the three-dimensional spectrogram graph of the data obtained from the guitar channel of the UAD Teletronix LA-2A software, a one-to-one simulation of the UAD company, is displayed. While the spectrogram graph of the analog device in Image 18 displays the color of the dynamic area of the signal more brightly, the data obtained from the software shows a more muted and loss of the natural tone of the signal. In the spectrogram graph of the analog device in Image 18, the saturation value is displayed in a distinct color temperature. In Image 19, in the simulation of the software model, this saturation value is observed to be at a lower temperature. In the data obtained from the software, it is observed that the saturation value is limited in the entire guitar channel, and the saturation ratio between 3 kHz and 20 kHz frequencies has significantly decreased.



Picture 20. Spectrogram graph of the guitar with the Insight 2 Pro plugin (Analog)



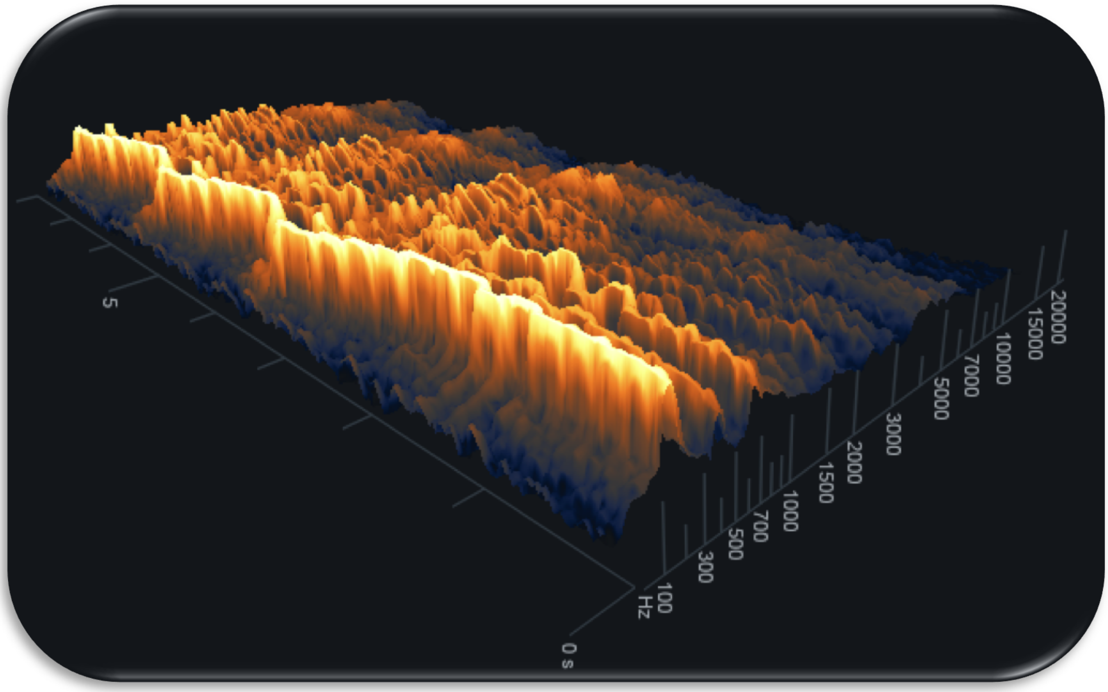
Picture 21. Spectrogram graph of the guitar with the Insight 2 Pro plugin (Software)

The three-dimensional spectrogram graph of the data obtained from the vocal channel of the Teletronix LA-2A analog device is shown in Image 20. In this analysis, a detailed topographic sound map of the vocal is created. In the spectrogram graph of the vocal channel in Image 20, after obtaining data from the Teletronix LA-2A analog device, the graph shape that the signal's static, dynamic, and natural tone should have over time is displayed.

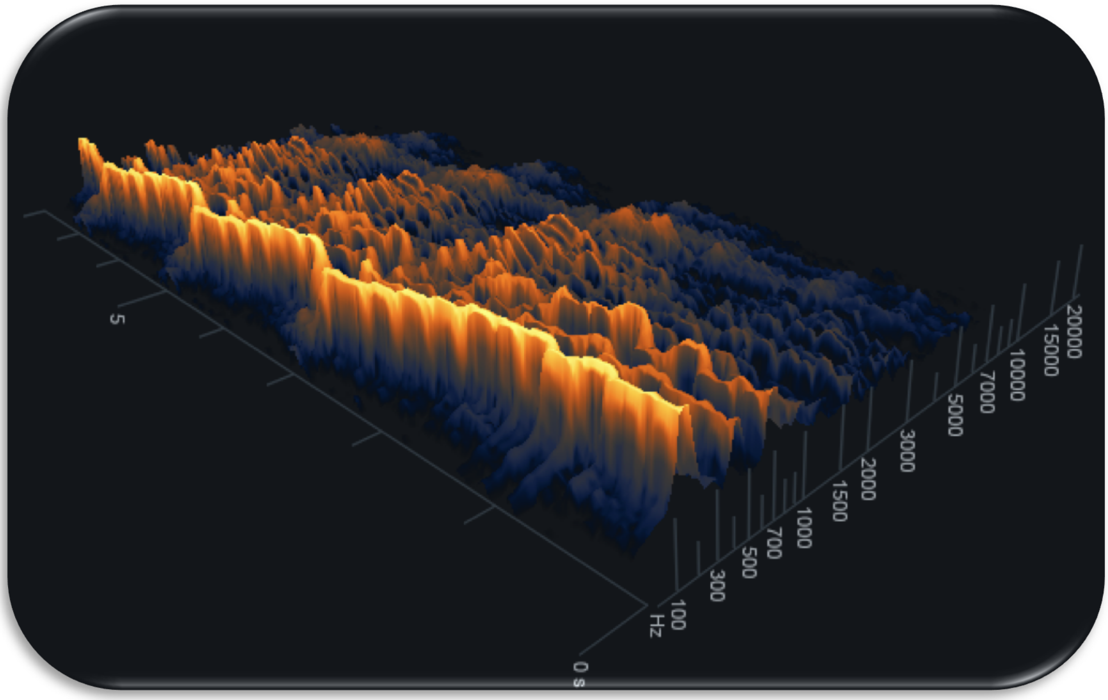
In Image 21, the three-dimensional spectrogram graph of the data obtained from the vocal channel of the UAD Teletronix LA-2A software, a one-to-one simulation of the UAD company, is displayed. While the spectrogram graph of the analog device in Image 20 displays the color of the dynamic area of the signal more brightly, the data obtained from the software shows a more muted and loss of the natural tone of the signal. In the spectrogram graph of the analog device in Image 20, the saturation value is displayed in a distinct color temperature. In Image 21, in the simulation of the software model, this saturation value is observed to be at a lower temperature. In the data obtained from the software, it is observed that the saturation value is limited throughout the vocal channel, and the saturation ratio between 1.5 kHz and 20 kHz frequencies has significantly decreased.

After obtaining data from the Teletronix LA-2A analog device, the file size of the guitar audio file changed from 2.48 to 2.46 megabytes, while after obtaining data from the software, the file size changed from 2.48 to 3.26 megabytes.

After obtaining data from the Teletronix LA-2A analog device, the file size of the vocal audio file remained constant at 4.14 megabytes, while the file size of the data obtained from the software changed from 4.14 to 5.49 megabytes.



Picture 22. Spectrogram graph of the vocal with the Insight 2 Pro plugin (Analog)



Picture 23. Spectrogram graph of the vocal with the Insight 2 Pro plugin (Software)

Conclusion

The analysis conducted in comparing the signals obtained from the Teletronix LA-2A analog device with those obtained from the UAD company's one-to-one simulation software of the Teletronix LA-2A indicates observed harmonic discrepancies. As a result of these evaluations, it has been observed that the software from UAD is less effective on harmonics compared to the Teletronix LA-2A analog device.

In the analog device, the master output peak level in the guitar channel was measured at 6.8 dB with a VU meter, while in the UAD simulation of the device, this ratio was observed to be -9.9 dB, indicating a difference. Through VU meter analysis, it was measured that there is a 16.7 dB difference in sound intensity loss between the analog device and digital simulation. In the data obtained from the vocal channel of the analog device, while the master output shows a level of 8.4 dB, the maximum peak level shown by the simulation of the device is -6.1 dB, revealing a difference. In the vocal channel, a sound intensity loss of 14.4 dB was observed between the analog device and its simulation.

In the spectrum analysis data, it was observed that the frequency components of the Teletronix LA-2A analog device and the UAD simulation software, which is a one-to-one simulation of the UAD company, are not in the same energy ranges. While the analog device does not apply any cutting operation in the low frequencies, the software simulation is observed to apply automatic cutting to frequencies between 20 Hz and 50 Hz.

The comparison of the data obtained from the Teletronix LA-2A analog processor and the UAD Teletronix LA-2A software resulted in the observation that the UAD software, by changing the dynamic range and noise level of the signal, does not reflect the same frequency response as the analog device. According to these analyses, it is evident that the Teletronix LA-2A digital software does not accurately reflect the characteristics of the analog processor in terms of imitation and realism. While saturation values vary depending on the working time of the Teletronix LA-2A analog processor, in the UAD Teletronix LA-2A software, saturation values are processed and set in a constant manner. The compression ratio does not change as the tubes heat up in the analog, unlike in the digital.

The Teletronix LA-2A analog processor operates independently without the need for a computer or a DAW interface. It has its own working principle. The UAD Teletronix LA-2A software, on the other hand, requires an interface to open this software via a DSP processor card and operates on a DAW software. Processing speed and response time vary depending on the processor speed of computers and DSP cards. The Teletronix LA-2A analog processor works independently without the need for a computer or DSP cards, so the processing speed and response time remain stable. While the Teletronix LA-2A analog processor is fixed and used in the studio, the UAD Teletronix LA-2A software is used anywhere without being fixed via a computer. There is no physical limitation. In addition, in terms of recording values and later working with the same values, the use of the Teletronix LA-2A analog processor is more practical and easier than the software. In terms of flexibility and adjustability levels provided by software in the mix process, parameter controls, effect options, and adjustments that can be made during processing are more practical compared to analog devices. At the same time, the effects of analog processors and digital software on the mix workflow vary depending on the number of devices used. While the parameter settings, memory placement, and access to pre-recorded settings for digital software are very easily provided, such a design interface is generally not found in signal processors as they are manual. In conclusion, DSP-based software processes sound recording data differently than analog devices. In this context, it has been concluded that the Teletronix LA-2A software, a one-to-one simulation of the UAD company, does not meet many features of the Teletronix LA-2A analog device, such as saturation, signal character, sound intensity, and frequency components.

Limitations of the Research

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

Acknowledgment

This study is derived from the doctoral thesis titled "A Comparison of Analog Processors and Digital Software" by the first author. The study has been approved by the Ethics Committee of the Department of Music, Institute of Fine Arts, Akdeniz University. No funding was received for the research, and there is no conflict of interest related to the subject.

Biodata of Author

Mehmet Özkeleş was born in June 1985 in Adana, Turkey. Coming from a musical family, Özkeleş has been immersed in music since his childhood. He completed his high school education at Anadolu Fine Arts High School, majoring in music, and simultaneously earned his undergraduate degree in Vocal Performance at Çukurova University State Conservatory. Following his successful high school years, Özkeleş continued his academic journey with a degree in Music Education at Harran University in Şanlıurfa, a Master's degree in Music Sciences and Technologies at İnönü University in Malatya, and a Ph.D. in Music Sciences at Akdeniz University. Mehmet Özkeleş, who has worked as a Lyric Tenor Soloist in various orchestras and has participated in numerous cultural and artistic events, continues his academic endeavors. In addition, he runs his own music studio where he creates his own compositions. Beyond his personal work, Özkeleş provides support in various fields such as arrangement, composition, production, and sound engineering to established and aspiring artists in the music industry.

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References

Bartlett, B. Bartlett, J. (2005). *Practıcal recording techniques*. Focal Press, USA, 211.

Burgess, J. R. (2014). *The history of music production*. Oxford: Oxford University Press, 30i

CNNTÜRK. (2008). Dünyanın en eski ses kaydı bulundu (World's oldest audio recording found). http://www.cnnturk.com/2008/yasam/diger/ 03/28/dunyanin.en.eski.ses.kaydi.bulundu/442466.0/index.html

Cumhuriyet Bilim Teknik, *No: 346*, 22 Ekim 1994.

Coşaner, N. (2008). *Volume. 48*, Haziran 77.

Durmaz, S. (2009). *Audio terimleri sözlüğü (Dictionary of audio terms)*. Cinius Yayınları.

Edstrom, B. (2011). *How to make great audıo recordıngs wıthout breakıng the bank.* Oxfor University Press, New York, 170.

Huber, D. M. (2010). *Modern recording techniques*. Focal Press, 199-200.

Izhaki, R. (2008). *Mixing audio concepts, practies and tools*, Focal Press, 365.

Lefford, N. M., (2000). *Recording studios without wall*. Master thesis. Massachusetts Institutes of Technology.

MC Queary, C. R. (1990). *The early years of the acoustic phonograph. its developmental origins and fall from* favor 1877-1929, A Senior Thesis In Historical American Technologies, 6, 11-16.

Mumma, G., Rye, H., & Kernfeld, B. (2022) Recording: I. Technological developments. Grove music online. Erişim: http://www.oxfordmusiconline.com:80/subscriber/article/grove/music/J371600

Öcek, C. (2010). *Elektro gitar efektleri ve amplifikatörler (Electric guitar effects and amplifiers)*. Pan Yayıncılık.

Önen, U. (2010). *Ses kayıt ve müzik teknolojileri (Sound recording and music technologies)*. Çitlembik Yayınları.

Pasinlioğlu, T./K. (2016). *Ses uygulamalarında efekt ve sinyal işlemciler (i Sound recording and music technologies)*. Cinius Yayınları.

Rumsey, F. (1994) *Midi systems & control*. Oxford: Focal Press, 77.

Schmidt-Horning, S. (2013). *Chasing sound: technology, culture, and the art of studio recording from Edison to the LP*. Baltimore: John Hopkins University Press, 36.

Yıldırım, A., & Şimşek, H. (2006). *Sosyal bilimlerde nitel araştırma yöntemleri (Qualitative research methods on social sciences). Seçkin.*