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Research Article

Implementation approaches and strategies in the use of equalizer in music production: A systematic review

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Abstract

The equalizer, a fundamental tool in the disciplines of music production and audio engineering, is one of the most important elements that shape the timbral qualities of sound, its perceptual clarity, and its hierarchical balance within the mix. With the proliferation of digital audio processing technologies, access to this tool has become easier, but this technological democratization has also brought about a serious competency problem. When inexperienced users, in particular, use equalizers without the necessary technical and theoretical background, this leads to fundamental production errors such as frequency masking, timbral muddiness, and the loss of natural qualities of the processed sound. The literature demonstrates that amateurs interested in the field experience cognitive difficulties in the face of the complexity presented by digital interfaces and tend to perceive equalizers as a magical tool promising instant solutions. The main motivation for this study is the paucity of academic studies that present a systematic synthesis of technical and practical information found scattered in the relevant literature and transform these elements into teachable pedagogical approaches. The aim of this research is to fill this academic-pedagogical gap. To this end, a systematic review model of qualitative research designs was designed based on the PRISMA 2020 guidelines. 49 academic and technical documents, selected based on a comprehensive literature review and predetermined criteria, were analyzed through content analysis. The analyses revealed that approaches to equalizer use in the literature can be structured under three fundamental frameworks: a corrective approach aimed at resolving technical issues, a shaping approach aimed at enriching the aesthetic character of sound, and a creative approach that transforms sound into a means of artistic expression. The practical application of each approach is illustrated with schematic diagrams. These frameworks are supported by key concepts from the literature, linking them to the "problem-solving" philosophies of experts such as Owsinski and Senior and the "tone perception" studies of researchers such as Dobrowohl. In conclusion, this research fills an important gap in the field by offering a holistic perspective that removes the use of equalizer, a multi-layered competency, from random trials and makes it more understandable, conscious, and teachable.

Keywords

equalizer, music production, music technology, signal processors, sound engineering

Introduction

Music production is a complex process that combines the artistic and technical aspects of sound, and one of the fundamental tools at the heart of this process is the equalizer. The basic function of an equalizer is to alter the frequency content of an audio signal by boosting or cutting specific frequency ranges. The equalizer plays a critical role in shaping the timbral character of the sound, its clarity, and the overall balance within the mix. While the proliferation of digital audio workstations (DAWs) has made this powerful tool accessible to users of all skill levels (Pakarinen et al., 2011; Ramola, 2022), this technological democratization has brought with it a new challenge: the inadequacy of pedagogical resources needed for a wide range of users to use these powerful tools consciously and effectively.

Unconscious interventions can negatively impact sound quality by causing serious technical problems such as frequency overlap, masking, and loss of naturalness in productions. Indeed, Mycroft and Paterson (2011) state that the complexity of digital interfaces creates cognitive difficulties for novice users, while Senior (2019) emphasizes that amateurs tend to view the equalizer as a magical "miracle drug" that provides an instant solution, which leads to erroneous interventions. This situation, combined with ever-rising listener expectations (Spotify Audio Labs, 2024), presents a significant competency challenge for both amateur and professional users.

The basis of this problem lies in the structure of the literature in the field. While numerous valuable resources exist (industry journals, professional blogs, books) that explain the technical principles and practical use of equalizers, these resources are generally in the form of "how-to" guides that teach the use of specific software or specific strategies for advanced professionals. The literature indicates that most existing guides target professional studios and ignore the practical issues faced by amateur users (Izhaki, 2008). However, there is a marked lack of academic studies that synthesize this scattered knowledge through systematic analysis and transform the underlying principles into teachable pedagogical models. This research aims to fill this academic-pedagogical gap.

This study offers unique value by reconstructing corrective, formative, and creative equalization approaches within the literature as replicable and teachable pedagogical practice methods within the framework of a systematic literature analysis. In line with this overall objective, the study seeks to answer the following fundamental research questions:

- ➢ What are the basic principles and systematic steps that constitute the corrective equalization approach used to solve technical problems in music production?
- ➢ What basic principles and practical strategies can be used to structure the shaping equalization approach to enrich the diagnostic and aesthetic character of sound sources?
- ➢ What basic techniques and application strategies are based on the creative approaches using the equalizer as a means of artistic expression?

The study, which seeks answers to these questions, first details the theoretical framework and research method of the subject, and then discusses these findings by presenting the findings and pedagogical models obtained from the literature analysis.

Theoretical Framework and Related Literature

The equalizer is considered a fundamental element of modern audio engineering and music production processes, playing a critical role in optimizing tonal balance, clarity, and aesthetic qualities by regulating the frequency spectrum of audio signals (Dewey, 2014; Rämö and Välimäki, 2014). The widely accepted notion in the literature emphasizes the important function of the equalizer in optimizing the sound quality and ensuring the integrity of the various sound elements within the mix (Owsinski, 2017). The use of an equalizer is not only a technical necessity but also functions as a creative instrument that determines the emotional and artistic impact of the sound. As Välimäki and Reiss (2016) point out, the equalizer is not only a tool based on objective principles of sound physics and signal processing, but is also considered a subjective field of application reflecting the sound engineer's aesthetic judgments, experience, and artistic vision. This dual structure explains the complexity of using the equalizer and why mastering it requires both technical knowledge and advanced listening skills (Corey, 2016).

History of the Equalizer

While the concept of equalizers is synonymous with modern electronics, the pursuit of understanding and consciously shaping the frequency content of sound dates back much further. For example, ancient theaters were structures "designed for perfect acoustics", and in these designs, each architectural element, such as the backstage wall or the orchestra area, had a special acoustic role in directing the sound (Bo et al., 2016: 83-84). This demonstrates that the effects of spatial designs on sound frequencies were already well understood in the past. The scientific manipulation of frequencies, however, was made possible in the 19th century by resonators, named after Hermann von Helmholtz and used to isolate specific frequencies (Sabat et al., 2022).

The first practical applications of frequency-specific signal manipulation emerged in the late 19th century with harmonic telegraphs designed in the 1870s. These systems, by sending multiple signals over a single line via vibrating reeds tuned to specific frequencies, laid the foundation for the modern idea of frequency-division multiplexing. Soon after, wave filters were used in telephone lines to compensate for high-frequency attenuation in long cables (Välimäki and Reiss, 2016: 83), a process called "equalization" (Önen, 2011).

With the proliferation of phonographs and gramophones in the early 20th century, "a standard mastering process was needed in which levels, equalizer settings and dynamics were adjusted" to ensure the consistency of recordings (Tanyeri, 2024: 15). The emergence of the first adjustable equalizers was accelerated by the sound industry's need to compensate for high-frequency losses in sound passing through perforated cinema screens. In this field, the external equalizer designed by John Volkman is considered one of the first examples, with selectable frequencies and cut-off features. These developments also carried over outside the studio, and following the bass and treble controls for gramophones in 1949, "potentiometer tone controls that give the user full control" were developed by Peter Baxandall in 1952 (Välimäki and Reiss, 2016: 83).

In the studio environment, equalizers became more sophisticated during the 1950s and 1960s, with pioneering designs such as the Langevin Model EQ-251A.

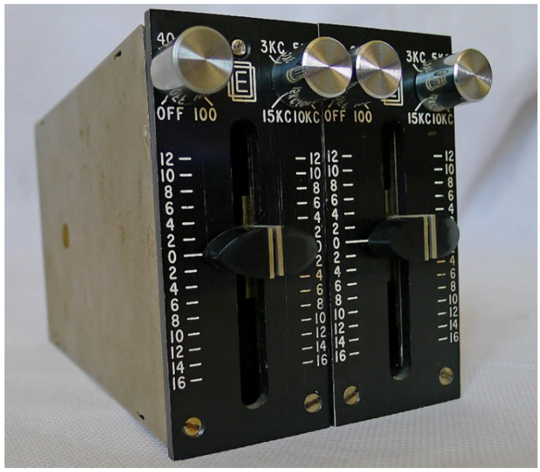

Photo 1. Langevin Electrodyne Model: 251A EQ-1956 (Web 1)

The invention of parametric equalizers in the 1970s, pioneered by engineers such as George Massenburg, laid the foundation for modern audio engineering by providing precise control over frequency, gain and bandwidth (Q).

Photo 2. The first parametric equalizer developed by Burgess MacNeal and George Massenburg. ITI Audio MEP-130 Console EQ-1969 (Web 2)

The popularization of digital equalizers, beginning in the 1980s, ushered in a new transformation in music production with groundbreaking features such as automation and dynamic control (Duggal, 2024). By the late 1990s, the global popularity of free, user-friendly media players such as Winamp had removed the graphic equalizer from the monopoly of professional studios. This became part of a broader consumer trend toward customizing audio experiences, allowing millions of home users to tailor audio frequencies to their personal preferences (Leyshon, 2009).

Nowadays, frequency and phase analysis can be performed automatically with artificial intelligence-supported equalizers, and more transparent and natural sound processing can be achieved thanks to advanced algorithms. These systems find the most appropriate equalizer settings with convolutional neural networks and genetic algorithms (Christensen, 2003; Engel et al., 2020; Başay et al., 2024), while preserving fine details in the signal and reducing noise with techniques such as MUSIC and Wiener filters (Proakis and Manolakis, 2007). Semantic control systems, particularly those that model human perception, produce natural results aimed at goals like “brightness” (Stasis et al., 2016), bringing together technical precision and artistic expression in music production.

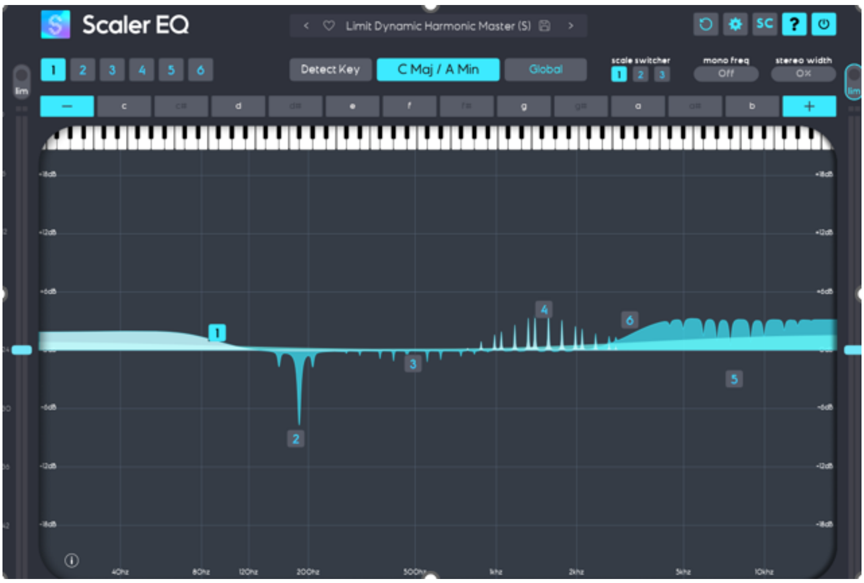

Figure 1. Plugin Boutige Scaler EQ

Psychoacoustic and Perceptual Effects

The use of an equalizer is not only a technical process that alters the physical properties of the audio signal, but also a psychoacoustic intervention that shapes the listener’s perceptual and aesthetic experience. Quantitative changes in the frequency spectrum create an effect that is defined as “perceptual features” in the literature and that translates into qualitative concepts such as “loudness” and “brightness” in the listener’s mind (Zölzer, 2011). As a matter of fact, it is stated that due to the natural sensitivity of the human ear, especially in the 1-5 kHz range, interventions made at these frequencies can make the music be perceived as “louder” or “stronger” (Tanyeri, 2024: 923). Therefore, sound engineers aim to create the desired timbral character and emotional effect by manipulating certain frequency ranges. This skill is a proven competence that can be improved in a short time with systematic training of the listener’s timbre discrimination ability (Corey, 2016).

It is accepted in the literature that interferences in certain frequency ranges create specific perceptual qualities. For example, it is stated that an increase in high frequencies causes music to “sound very ‘bright’ and ‘scratchy’” (Rumsey and

McCormick, 2009: 331). Similarly, it is stated that the sensation of "warmth" and "body" arises from the balanced distribution of the energy constituting the timbre in the lower and lower-mid frequencies (Tanyeri, 2024). Uncontrolled accumulation in these areas leads to a problem called "muddiness", where clarity between frequencies is lost; It is emphasized that in cases of excessive intensity, especially in the 100-400 Hz range in vocals, "the vocal sound loses its clarity and turns into a heavy and muffled sound like mud" (Kim, 2018: 58).

A "muddy" sound in modern music production stems not only from flawed mixing decisions but also from an industrial trend known as the "Loudness War," which has led to music recordings "increasingly containing compressed, loud, and static sounds" (Tanyeri, 2024: 50). Narrowing the dynamic range in an attempt to sound louder eliminates the nuances and details within the music, reducing the separation between frequencies and causing an overall monotony. It is also stated that the characteristic sound of modern recordings is "compression" and that the effects it creates, such as phase shift, are now considered natural, but those who defend traditional sound aesthetics are uncomfortable with this situation (Owsinski, 2017). Addressing this issue and preserving the timbral integrity of the sound relies on critical listening skills that mastering engineers develop through years of perceptual training (Corey, 2016).

Equalizer working principle and basic components

The operating principle of an equalizer is based on fundamental concepts such as frequency, harmonics, and filters. Frequency is the fundamental parameter that determines the pitch of sound and is measured in Hertz (Hz). The human ear can generally hear frequencies between 20 Hz and 20 kHz. Equalizers divide this frequency range into different bands and offer the option of controlling each band separately (Pasinlioğlu and Pasinlioğlu, 2016). While analog equalizers provide a warm and organic auditory experience due to distinctive harmonic distortions and phase deviations, digital equalizers provide precise, adaptable and repeatable frequency control (Välimäki and Reiss, 2016). Understanding these fundamental components is crucial for informed equalizer use.

Frequency ranges and diagnostic implications

To understand the functions of the filters used in equalizers, it's first necessary to define their frequency ranges. In music production, the 20-20,000 Hz frequency range, which represents the human hearing threshold, is generally examined under six headings: Sub-Bass, Bass, Low-Mid, Mid, High-Mid, and High.

Audio frequencies are important components that shape the overall character of music and sound. The sub-bass (20 Hz-60 Hz) range provides the perceived power and depth of a sound; boosting in this region adds weight and depth, while attenuation prevents unnecessary rumble and makes the sound lighter. The bass (60 Hz–250 Hz) provides the rhythmic foundation and fullness of the music; boosting provides a full and powerful sound, while attenuation improves clarity and reduces low-frequency buildup. Lower-mid (250 Hz–500 Hz) frequencies affect the warmth and body of instruments; excessive boosting can result in muddy or muffled sounds, while balance with sub-bass frequencies is important. The mid-range (500 Hz–2 kHz) range determines the fundamental tonal character of the human voice and many instruments and is critical to sound clarity. Upper-mid (2 kHz–4 kHz) frequencies increase sound detail and clarity; While boosting makes the sound "lively" and "clear," excessive boost can create a harshness that tires the ear. The upper frequencies (4 kHz-20 kHz), on the other hand, provide brightness, clarity, and clarity. In particular, between 4 kHz and 6 kHz, it increases the expressive power of the sound, while between 6 kHz and 20 kHz, it adds "sizzle" and realism to the sound; however, excessive boost can create a "brightness" or "hiss" that tires the ear (Alm and Walker, 2002; Corey, 2016; Loni, 2013; Özkeleş and Arapgirlioğlu, 2024).

In music production, understanding and interpreting the frequency ranges of instruments and human voices is crucial for correctly determining which interventions to perform at which stage using equalizers. Figure 2 shows the fundamental frequency ranges and harmonic boundaries of instruments and human voices.

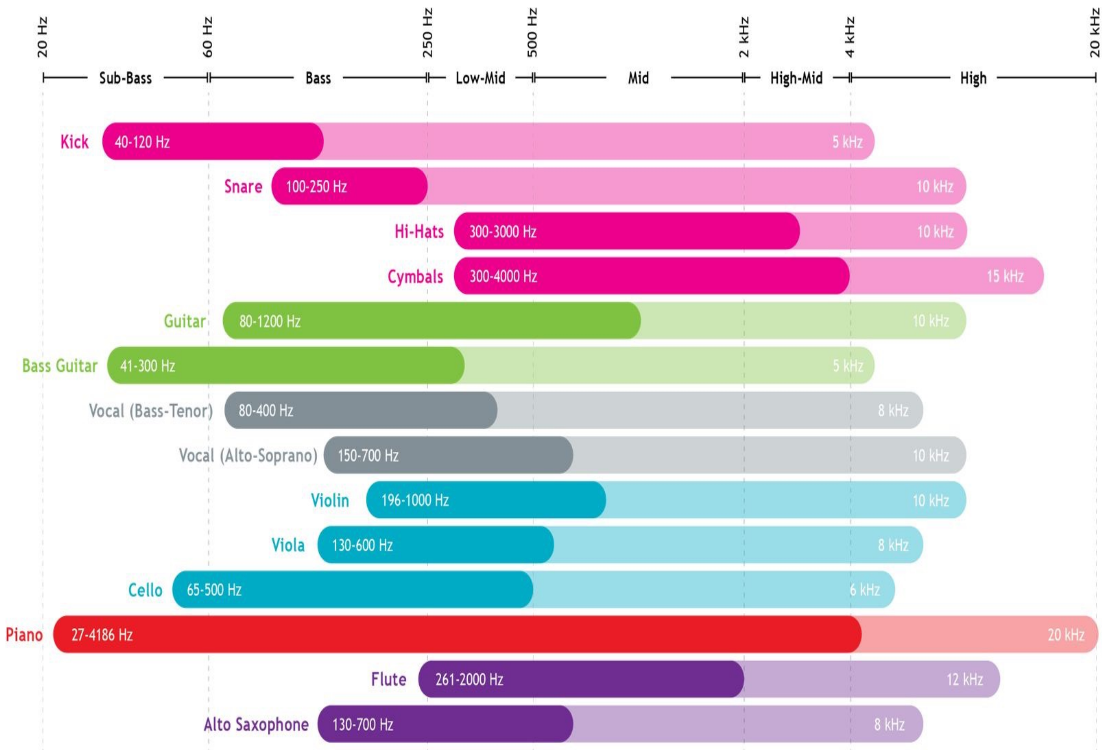

Figure 2. Instrument and human voice frequency ranges (adapted from Owsinski, 2017; Senior, 2019)

Filter types

Filters, the fundamental building blocks of equalization, offer different characteristic curves to precisely shape specific sections of the audio spectrum. These filters are divided into basic categories: frequency-pass filters (HPF/LPF), which primarily clean unwanted frequency ranges; band-pass filters (Band-Pass/Notch), which isolate or eliminate a specific frequency band; and shelf filters, which adjust the overall tonal balance of a sound (Proakis and Manolakis, 2007; Rumsey and McCormick, 2009; Zölzer, 2011). In addition to these basic filters, the peaking filter, which allows for surgical interventions on the timbral characteristics of a sound, and the advanced all-pass filter, which alters the phase relationships of a signal, also hold an important place in modern production techniques (Dutilleux et al., 2011, p. 52). Each filter has its own definition, function, and primary application. These basic filter types, commonly used in music production, are summarized in detail in Table 1.

Table 1. Basic filter types and functions used in equalizer

Filter Type	Filter Name	Definition and Function	Basic Usage Area
Frequency Pass Filters	High-Pass (HPF) Low-Pass (LPF)	HPF: Cuts below the specified frequency and passes above it. LPF: Cuts above the specified frequency and passes below it.	Cleaning of unnecessary low-end noise (HPF) or high-pitched hum (LPF).
Bandpass Filters	Band-Pass and Notch	Band-Pass: Passes only a narrow range of frequencies. Notch: Cuts only a narrow range of frequencies.	Creating special effects (Band-Pass), eliminating a certain resonance or noise (Notch).
Shelf Filters	High-Shelf Low-Shelf	Collectively boosts/lowers all frequencies above (High) or below (Low) the specified frequency.	Adjusting the overall tonal balance, adding an overall brightness or fullness to the sound.
Peak Filter	Peak (Bell)	It precisely amplifies or attenuates a bell-shaped area around a center frequency.	Emphasizing certain instrument timbres or controlling problematic resonances.
All Pass Filter	All-Pass	It only changes the timing (phase) of the frequencies without changing their volume (amplitude).	Correcting phase shift issues and advanced sound design applications.

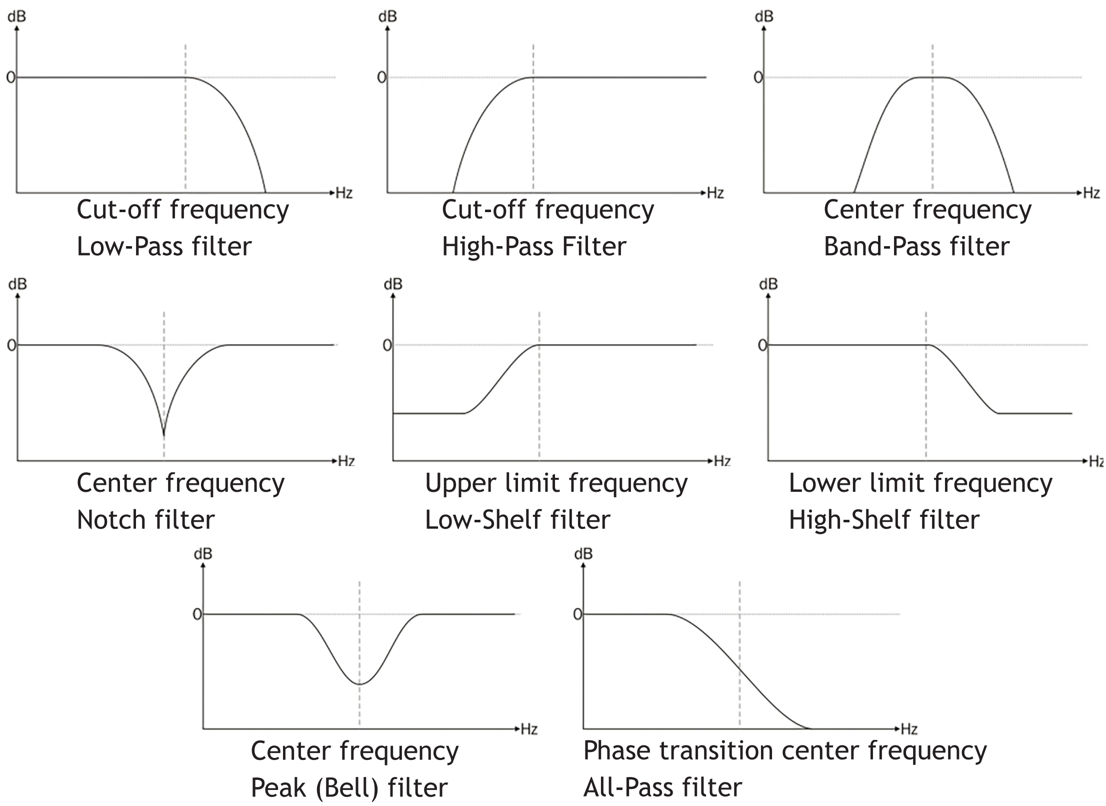

Figure 3. Equalizer filter graphs

Equalizer types

Various equalizer types are used in music production, each offering different levels of control and flexibility. Graphic equalizers, distinguished by their visual interfaces for live sound and quick tonal adjustments (Rämö et al., 2014; Rämö and Välimäki, 2014); Parametric equalizers, which offer the most flexible and surgical control in the studio environment (Välimäki and Reiss, 2016); and Dynamic equalizers, which intelligently respond to the dynamics of the audio signal and offer solutions where traditional equalizers fall short (Owsinski, 2017; Stasis et al., 2016), are cornerstones of today's production systems. These types differ based on their adjustable parameters and basic operating logic, providing the sound engineer with a wide range of tools for specific purposes. The most common basic equalizer types, their features, and common areas of use are presented in Table 2.

Table 2. Basic features of equalizer types

Equalizer Type	Basic Features	Main Areas of Use
Shelving Equalizer	It shapes the frequency ranges in the audio signal that are below (Low Shelf) or above (High Shelf) a specific cutoff frequency. It is used to broadly adjust the overall tonality of the sound.	It contributes to the perception of spatial distance in the mix.
Graphic Equalizer	It consists of a series of peaking filters with fixed center frequencies and bandwidths. The gain of each frequency band can be adjusted independently.	Ideal for precisely targeting specific resonances or sculpting the tone of instruments in detail.
Parametric Equalizer	It provides full control over one or more frequency bands. Users can freely adjust the center frequency, gain, and bandwidth (Q factor) for each band.	Ideal for precisely targeting specific resonances or sculpting the tone of instruments in detail.
Semi-Parametric Equalizer	It is a simpler version of the parametric equalizer. It usually only offers center frequency and gain control, while the bandwidth setting remains fixed.	Used for simple and quick frequency and gain adjustments.
Dynamic Equalizer	It automatically adjusts the gain of specified frequency bands based on the level of the audio signal. Parameters are adjusted over time based on signal characteristics.	It is a powerful tool for resolving frequency conflicts in the mixer and improving signal clarity.

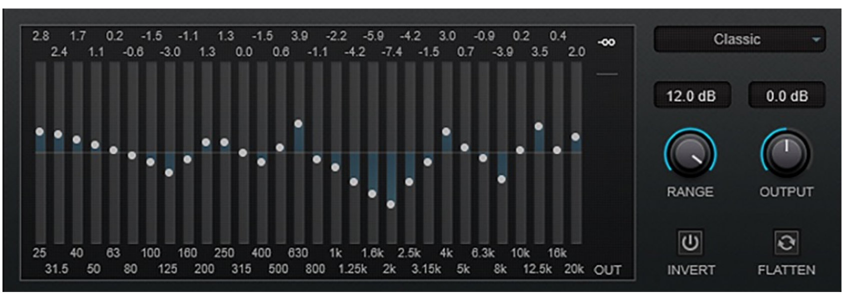

Figure 4. Graphic equalizer (Steinberg Cubase Pro 14. GEQ-30)

Figure 5. Parametric equalizer (Steinberg Cubase Pro 14.Studio EQ)

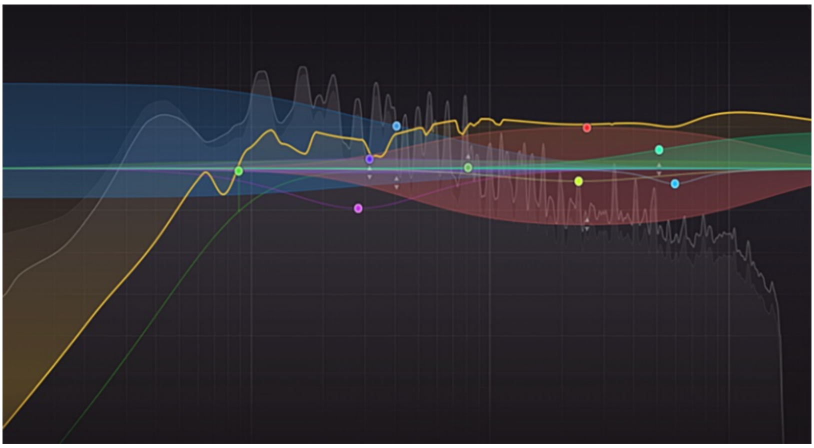

Figure 6. Dynamic equalizer (Plugin Boutique. FabFilter Pro-Q 4)

Equalizer parameters

Equalizers use several basic parameters to precisely shape the frequency spectrum of audio signals. Frequency determines which range of sound to intervene in, while gain adjusts the strength of this intervention (Kim, 2018; Başay et al., 2024). The most important complement of these two parameters, bandwidth (Q Factor), determines how wide or narrow the effect will be around the selected center frequency. This control allows sound engineers and producers to manipulate frequencies in more detail (Childs, 2012, as cited in Aras and Temuçin, 2022, p. 130). In addition to this basic trio, other critical settings include the slope, which determines the filter's characteristics, and the cutoff frequency, which limits its effect (Dewey, 2014; Kim, 2018). These key parameters and their functions, which form the basis of equalizer use, are explained in detail in Figure 7 and Table 3.

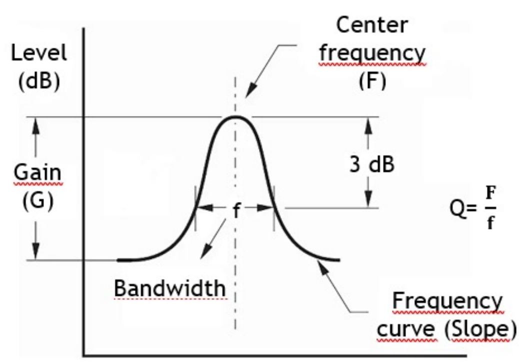

Figure 7. Equalizer parameters

Table 3. Functions of equalizer parameters

Parameter	Definition and Function	Effect
Frequency	The center frequency determines the filter's effect. The filter is effective not only at this frequency but also at frequencies surrounding it.	Defines the frequency region (bass, middle, treble, etc.) to be intervened.
Gain	Determines how much the intensity of the audio signal in the selected frequency band will be increased (boost) or decreased (cut).	Increases or decreases the volume of specified frequencies.
Bandwidth / Q Factor	It determines the width of the filter's response. The Q factor controls how narrow or wide this frequency range is, and therefore the filter's frequency selectivity.	Narrow bandwidth: Ideal for troubleshooting or fine-tuning specific frequencies. Wide bandwidth: Better for general tone shaping or more general adjustments.
Cutoff Frequency	The frequency at which a filter begins to pass or block signals is the point. 3dB, used as the cut-off frequency for a signal, represents the point at which signal strength is halved.	It allows you to reduce unwanted frequencies or emphasize certain frequency ranges.
Slope	It refers to the slope of the filter at the edges of the frequency band. This slope determines how smooth or sharp the filter transitions across the frequency spectrum.	Higher dB/octave values represent sharper slopes, while lower values represent smoother transitions.

Equalizer application approaches and current discussions in the literature

In the literature, equalization application approaches are generally classified under three main headings: corrective, shaping, and creative (Aras and Temuçin, 2022; Kim, 2018; Stasis et al., 2016). Current technological advances, particularly artificial intelligence (AI) and machine learning-based "smart" equalization systems, play a significant role in the implementation of these approaches. These systems can optimize the sonic quality of a musical piece based on ideal frequency spectra learned from commercial recordings and prevent the overcompression problem known as the "Loudness War" by automatically adapting a track to the loudness standards of different platforms (Spotify -14 LUFS, Apple Music -16 LUFS) (Tanyeri, 2024).

At the heart of these capabilities lie Digital Signal Processing (DSP) techniques, which separate signals into frequency components using algorithms such as the Fast Fourier Transform (FFT), and machine learning architectures such as Deep Neural Networks (DNNs), which interpret this data and estimate parameter settings (Oppenheim et al., 1999). More advanced models, such as Generative Adversarial Networks (GANs), are opening new horizons by enabling these systems to undertake creative mastering tasks (Canyakan, 2025). However, despite these technical capabilities, it is also emphasized that current systems cannot fully emulate a sound engineer's artistic vision and context-specific aesthetic decisions. While AI models generally aim for technical "accuracy," they are limited in understanding the cultural nuances required by the music, unexpected creative preferences, or the artistic goals of a project (Canyakan, 2025). Therefore, current research in the field investigates more integrated and user-centered approaches to overcome these limitations, such as systems that personalize voice based on biometric data and interfaces with haptic feedback.

Method

This section includes information about the research model, data collection process and data analysis.

Research Model

This study utilized the systematic review model, a qualitative research design. A systematic review is a qualitative research design that aims to integrate existing knowledge in a specific field with a structured methodology and subject it to critical evaluation. Within this model, relevant literature is scanned according to predetermined inclusion and exclusion criteria, the methodological quality of selected studies is meticulously examined, and the findings are synthesized (Karaçam, 2013). This process, as emphasized by the American Psychological Association (APA, 2020) for review articles, not only summarizes published material but also offers an original contribution to the field by organizing and integrating existing knowledge. Accordingly, the current study adopted this methodological framework. The primary purpose of this model is to systematically analyze the extensive and fragmented literature on equalizer use and to integrate theoretical knowledge in this field with practical applications. In this context, the research goes beyond summarizing the existing literature and aims to synthesize the analyzed approaches. The final finding of the study, based on this synthesis, is to present a structured framework that shows how theoretical approaches are embodied in practical scenarios. PRISMA 2020 (Preferred Reporting Items for Systematic Reviews and Meta-Analyses) guidelines were followed in the planning and reporting of the study.

Eligibility Criteria

Source selection for this systematic review was based on predetermined inclusion and exclusion criteria. Studies for inclusion were selected based on the following criteria:

Inclusion Criteria

For a study to be included in the review, it must meet all of the following criteria:

Publication Type: The study must be published in one of the following types:

- Research article published in a peer-reviewed journal
- A scientific book or book chapter
- Postgraduate thesis (PhD, Master's)
- Academic conference paper (Audio Engineering Society, DAFx)
- Online articles or product manuals that provide specific technical information from authoritative industry sources.

Subject Scope: The study addresses the use, techniques, principles, or historical development of equalizers directly in the context of music production, audio engineering, or mixing.

Language: The main text of the study must be in English or Turkish.

History: To ensure coverage of primary sources covering the historical foundation and classical approaches to the subject, no initial date limitation was applied. However, to capture modern approaches and current debates, special priority was given to works published within the last 10 years.

Exclusion Criteria

Publications with at least one of the following characteristics were not included in the qualitative synthesis phase of the review:

- Informal publications (general blog posts, forum discussions, popular magazine articles) that do not carry academic or verifiable technical reference value.

- Studies that focus solely on abstract signal processing theory, without any applied examples.
- Publications for which full text is not available.
- Personal video transcripts that do not offer a repeatable methodology based on the transfer of opinion or experience.

Information Sources and Search Strategy

A comprehensive, multi-stage literature review was conducted to identify the publications that comprise the research dataset. The review process utilized international and national academic databases such as Google Scholar, Scopus, Web of Science, AES e-Library, DergiPark, and ULAKBİM TR Index, as well as Scispace software and Niğde Ömer Halisdemir University Library resources. In addition to academic resources, industry publications such as Sound on Sound and studies by qualified professionals in the field were also reviewed to understand current industrial applications and practical discussions on the topic. However, only studies that met the eligibility criteria defined in the previous section were included in the final analysis.

The search strategy is structured to include Turkish equivalents of key keywords such as “equalizer”, “music production”, “audio mixing”, “frequency masking”, “audio engineering” and “signal processors”. The search was further deepened with more specific term groups such as “equalizer filter types,” “equalizer in mixing stages,” and “equalizer parameters.” These terms were systematically combined using Boolean operators and other filtering techniques (“ “, -”) to broaden or narrow the search scope. Furthermore, by utilizing AI-supported literature search tools such as Scispace, queries using natural language sentences and advanced filtering options (year, subject, similar publications, etc.) were used beyond keyword-based searches. Finally, to supplement database searches, additional sources were accessed using the “snowballing” method by examining the bibliographies of key studies.

Data Collection Process

The research dataset consists of academic, scientific, and technical documents on equalizers and music production. Document analysis was used as the primary method in the data collection process. Studies identified through the literature review were subjected to a four-stage selection process, following the PRISMA 2020 flowchart presented in Figure 8. In the first stage, records from various sources were combined, and duplicates were eliminated. In the second stage, the titles and abstracts of the remaining studies were screened according to the eligibility criteria, and irrelevant ones were excluded. In the third stage, the eligibility of the studies reviewed in full text was reassessed. In the final stage, the studies to be included in the qualitative synthesis were identified.

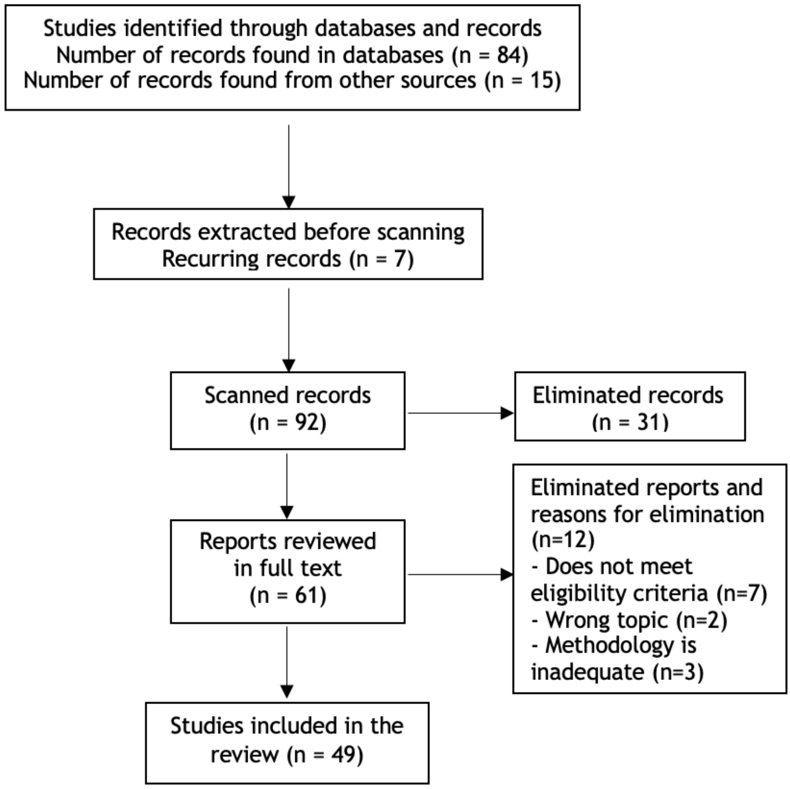

Figure 8. PRISMA 2020 flow chart

Data Analysis

Data obtained from the documents included in the compilation, in accordance with the PRISMA flowchart, were analyzed using content analysis, a qualitative research method. The analysis process included the following steps to ensure a systematic structuring of the findings:

Coding: All studies included in the compilation were carefully read and coded by identifying meaningful sections and concepts such as the principles of equalizer use, practical strategies, technical approaches and perceptual dimensions.

Creating a Theme: The codes generated in the first stage were grouped and compared according to their relationships. This process identified high-level patterns that recur in the literature and reflect different purposes for using equalizers.

Structuring Approaches: In the final phase of the analysis, the identified themes were synthesized to structure three fundamental application approaches that answer the fundamental research questions and generate its findings. These approaches are as follows:

- ➢ Corrective approach aimed at eliminating technical problems,
- ➢ A shaping approach that aims to enrich the aesthetic character of the sound,
- ➢ A creative approach that transforms sound into a means of artistic expression.

This synthesized thematic structure enabled the transformation of scattered information into concrete and applicable findings, which are the original contribution of the research.

Findings

A systematic literature analysis revealed that approaches to equalizer use in the literature can be structured under three fundamental approaches: corrective, formative, and creative. These structured models, the core findings of the research, transform the field's scattered theoretical knowledge and practical strategies into a concrete and teachable framework. The implementation steps for each approach are illustrated with schematic diagrams illustrating the underlying principles.

Corrective equalizer implementation approach: Troubleshooting technical issues

Content analysis of the literature included in this review reveals that the first and most fundamental approach to using equalizers is corrective or surgical. This approach focuses on resolving specific spectral issues found in audio channels (Dewey, 2014). Mastering engineers specifically use digital equalizers for this "surgical" purpose to eliminate problem areas such as "unwanted frequencies" and "anomalies" (Nilsson, 2024). The primary goal of these interventions is to control frequency responses that affect the "tonal quality and clarity" of the audio (İmik and Uçar, 2024). Strategies to address issues such as unwanted noise, resonance, and "muddying" include techniques such as filtering high frequencies for sibilance or scanning for resonant frequencies and reducing their gain (Oppenheim et al., 1999). Strategies to prevent frequency masking include more advanced methods such as dynamic equalizers, multi-band dynamic processors, and cross-adaptive architectures to ensure inter-channel spectral balance (Uriostegui-Hernandez et al., 2025). The corrective application approach developed from this synthesis demonstrates how these principles can be applied in practice through a concrete scenario.

Aim: To obtain a clean and processable audio signal for subsequent production stages by removing unwanted background noise from the microphone signal.

Scenario: A microphone signal is detected with a distinct "hum" noise around 60 Hz, presumably from the power grid, and a general hum in the lower regions of the spectrum. This is represented by a narrow, distinct peak around 60 Hz, shown schematically in Figure 9.

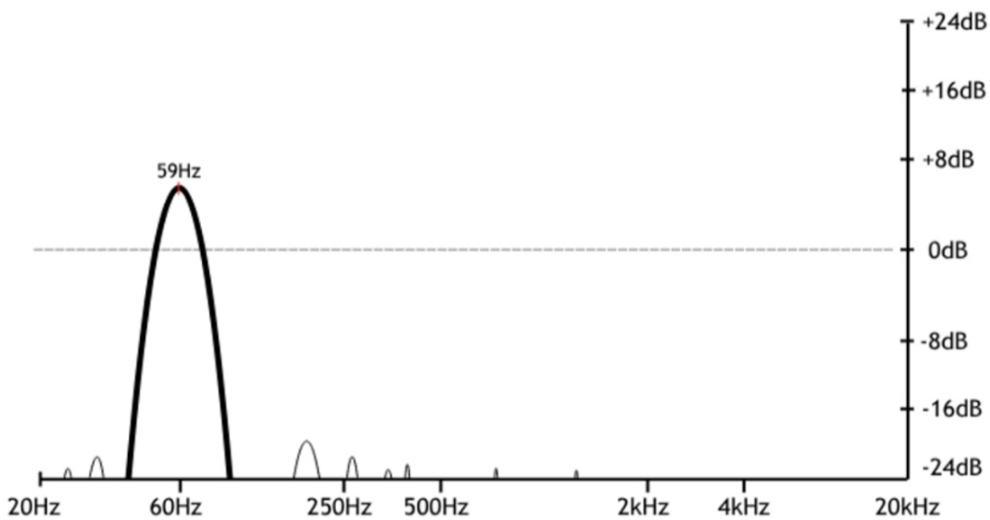

Figure 9. Signal spectrum before correction

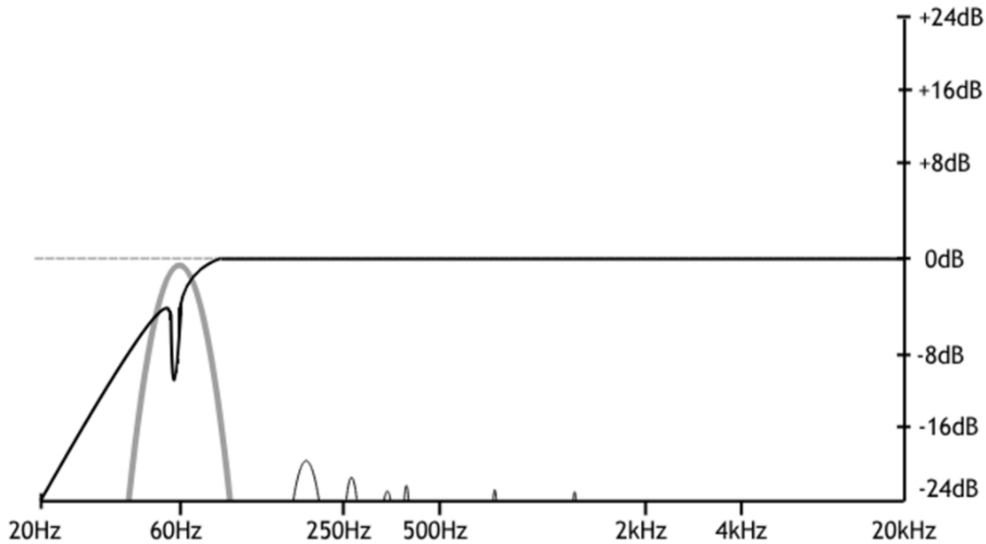

Figure 10. Corrective equalizer application

Approach to applying a shaping equalizer: Enriching the timbral character

Literature analysis reveals the existence of a shaping application in which the equalizer is used for aesthetic purposes. At the heart of this approach is the idea that timbre shaping plays as important a role as pitch and loudness in modern music production, and that listeners' judgments of a mix are largely based on timbre perception (Corey, 2016; Dobrowohl et al., 2019). In this approach, equalization is used for artistic purposes, such as controlling the timbral balance of the music (Aras & Temuçin, 2022), refining vocal tone (Kim, 2018), or, as George Massenburg puts it, "refining and maturing tones" (Owsinski, 2017), rather than solving a technical problem. Translating these subjective goals into technical parameters has also been the focus of academic research aimed at mapping perceptual descriptors like "warm" or "treble" to specific equalizer settings (Cartwright and Pardo, 2013). In professional practice, this translates into specific strategies, such as using the 10-15 kHz range to add "air" to the sound, or lower-mid frequencies like 300-500 Hz to add "warmth" and "body" (Owsinski, 2017). The sculpting equalizer approach is explored below through the scenario of making a vocal recording more prominent in the mix.

Aim: To enrich the aesthetic and timbral character of a vocal lost among instruments by increasing its clarity, brightness and presence in the mix.

Scenario: The production process addressed a vocal signal that was recorded cleanly, but lacked clarity and brilliance among the other instruments in the mix. This situation represents a need for timbral balance rather than a significant technical flaw and is schematically illustrated in Figure 11.

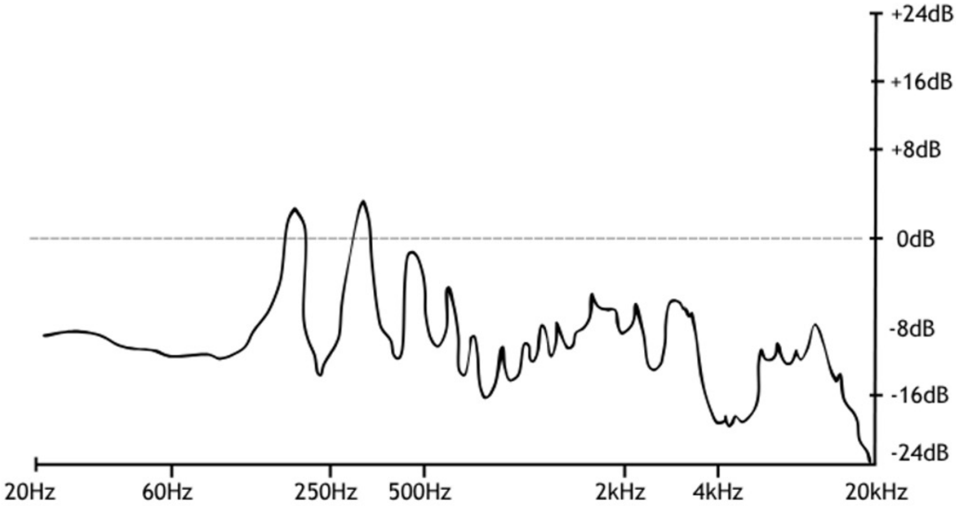

Figure 11. Vocal signal before shaping

Sample Application: In this scenario, the shaping approach consists of three complementary steps. The first step is to apply a gentle high-pass filter at approximately 100 Hz to remove unnecessary low-frequency intensity from the vocal. The second step is to apply a gentle +3 dB gain increase with a wide-band (low-Q) Bell filter in the 2-5 kHz range to increase the vocal's intelligibility and presence in the mix. Finally, a gentle +2 dB gain increase with a gentle high-shelf filter at 8 kHz and above is added to add a sense of "brightness" and "air" to the sound. The final shaping equalizer curve resulting from the combination of these three processes is schematically presented in Figure 12.

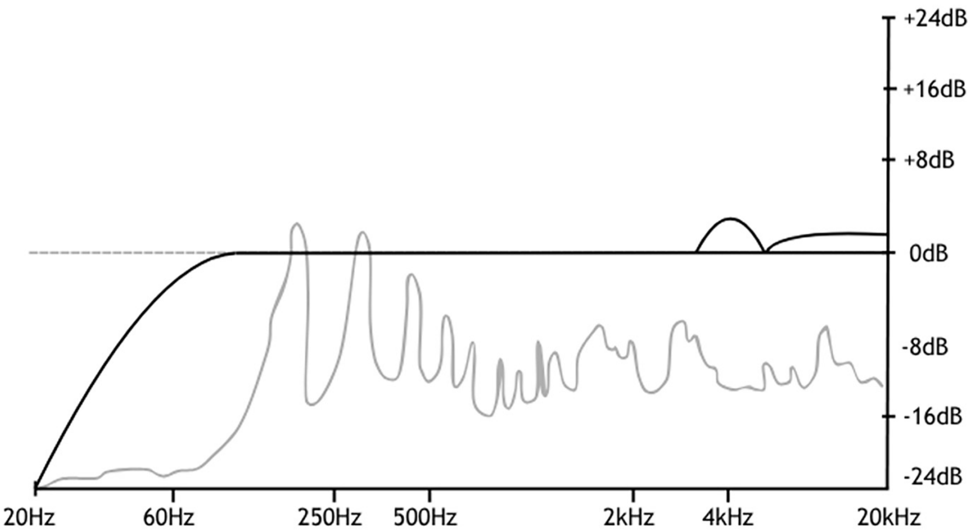

Figure 12. Shaper equalizer application

Creative equalizer application approach: Use as a means of artistic expression

The analyzed literature reveals the existence of a third approach, where the equalizer is used as a creative effect and sound design tool that goes beyond problem-solving and aesthetic enrichment, and radically transforms the sound. In this approach, the aim is not to achieve a natural timbre, but to consciously manipulate the sound spectrum to achieve a desired new timbral aesthetic (Stasis et al., 2016). This philosophy manifests itself in the "Mixing as a Performance" approach, where some sound engineers view the mixing process as a "performance" and use tools such as equalizers for artistic expression and improvisation. In this context, engineers such as Lee DeCarlo use the equalizer to "add" something new to the sound rather than "remove" something, while Tony Maserati makes equalizer decisions according to the energy of the song and the story it tells (Owsinski, 2017). The most well-known examples of this creative use in practice are; This can occur by using a narrow band-pass filter to create a telephone or radio effect, or by moving the filter's center frequency to achieve dynamic effects such as "filter sweep" and "wah-wah" (Dutilleux et al., 2011). In advanced applications, the character of the sound can be transformed into a completely unique and hybrid structure using techniques such as "vocoding" or timbral transformation (Verfaille et al., 2011).

Aim: To provide an atmospheric change by creating a nostalgic AM radio effect during certain sections of a song.

Scenario: The goal is to create an atmospheric change in certain sections of a fully mixed, full-spectrum piece, such as the intro or bridge. The spectrum of the unaffected signal is shown in Figure 13.

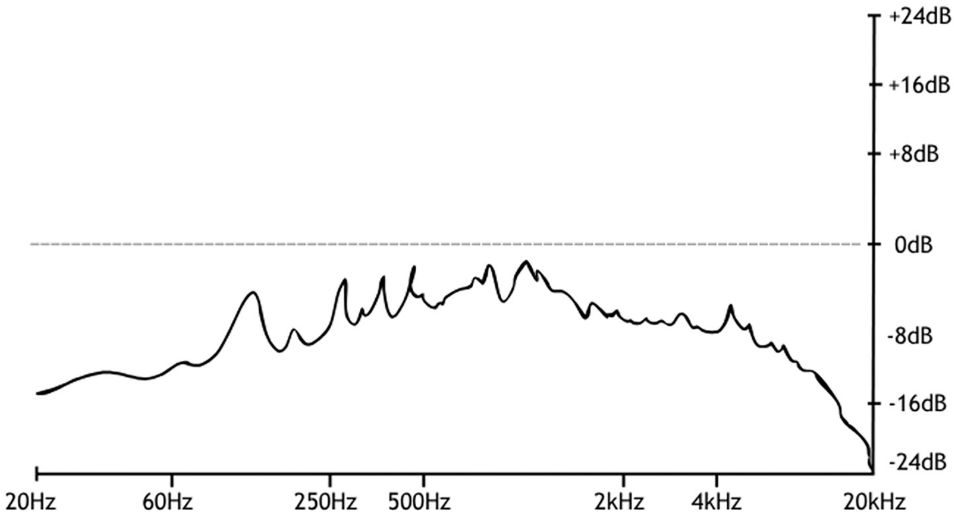

Figure 13. Mixed signal before application

Sample Application: In this scenario, the creative approach combines multiple advanced audio processing techniques. First, a steeply sloped Band-Pass filter is applied to mimic the narrow frequency range of AM radio broadcasts, cutting off the entire spectrum except for sounds between approximately 300 Hz and 4 kHz. To enhance the effect's character and mimic the mono nature of older radios, this filtering is applied only to the signal's mid-channel. Furthermore, a Dynamic Equalizer, which responds to the energy of the mix, adds a slight emphasis to the mid-frequencies, adding vibrancy to the effect. The equalizer curve with this radical filter is shown in Figure 14.

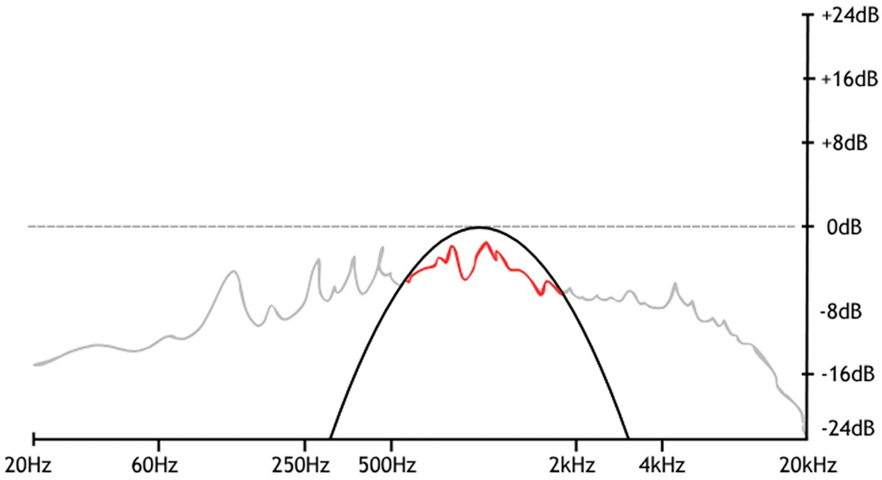

Figure 14. Band-Pass filter application for “Radio Effect”

Finally, to ensure this effect is heard only in the desired sections, rather than throughout the entire song, the “Bypass” and “Volume” automations are used in the DAW. This time-based implementation is schematically illustrated in Figure 15.

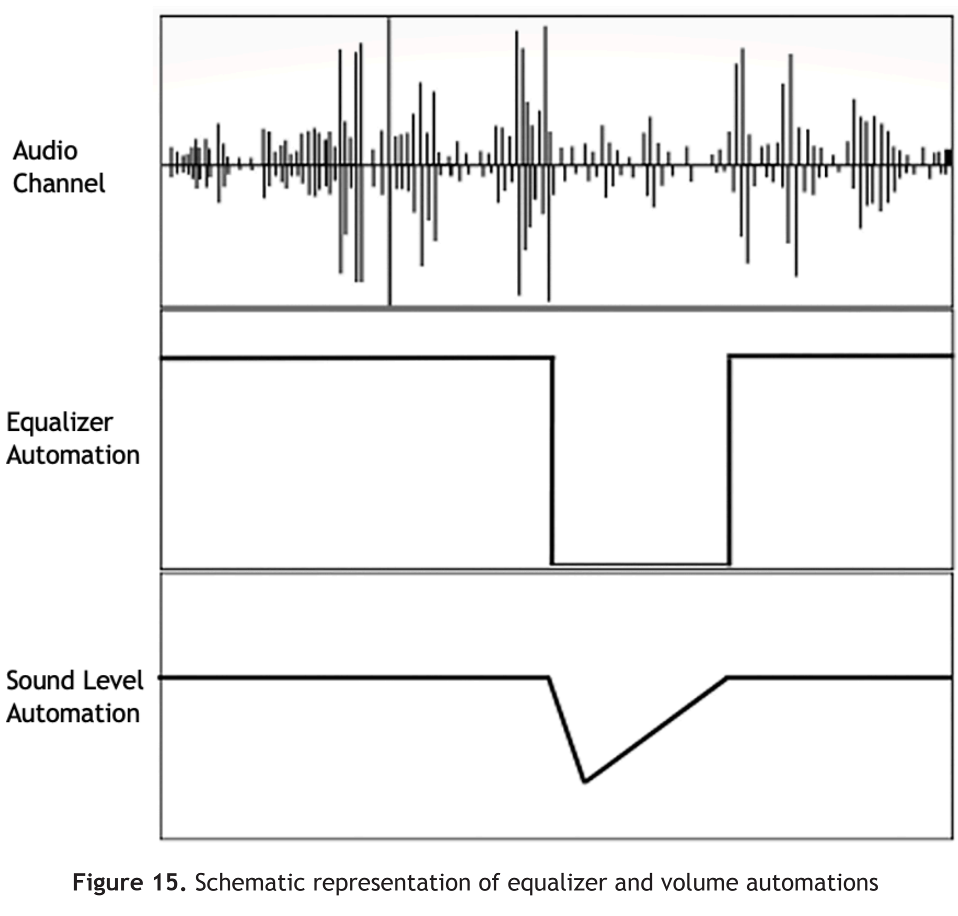

Figure 15. Schematic representation of equalizer and volume automations

Discussion

The key finding of this study is that the use of equalizers in music production can be categorized into three systematic approaches: corrective, formative, and creative, synthesizing scattered information in the literature. This classification offers a conscious and targeted methodology to address the problem of "unconscious use," which has become particularly prevalent with the democratization of technology (Mycroft & Paterson, 2011) and the tendency of amateurs to view equalizers as a magical "miracle drug" (Senior, 2019). This study aims to transform the use of equalizers from a haphazard trial-and-error process into a teachable process by not only naming the approaches but also turning them into structured frameworks that include specific purposes, scenarios, and process steps.

The corrective approach synthesized in this research is based on the "problem-solving" philosophy emphasized in the literature by experts such as Owsinski (2017) and Senior (2019). This philosophy dictates that the primary function of an equalizer is to eliminate frequency imbalances in an audio signal and solve specific sonic problems. These problems range from simply eliminating acoustic feedback (Rämö and Välimäki, 2014) to compensating for the loss of dynamic nuance caused by the "Loudness War" trend (Tanyeri, 2024). At the heart of this approach lies the advanced auditory evaluation skills of audio engineers, defined as "critical listening," developed over years of practice (Corey, 2016; Owsinski, 2017). This "problem-solving" process, often described empirically in the literature, is concretized by breaking it down into systematic and repeatable steps, as presented in the "Findings" section. This structured framework offers an important pedagogical solution to the problem of "unconscious use" in the field by providing a clear and well-founded methodology that both novice users and students can follow.

The shaping approach framework presented in this study demonstrates the aesthetic and artistic potential of the equalizer. The theoretical basis of this approach is the idea that listeners' judgments of a mix are largely based on timbre, and that timbre shaping plays as important a role as pitch and loudness in modern music production (Dobrowohl et al., 2019). Neuroscientific studies have also shown that timbre manipulations can evoke tangible physiological and emotional responses in listeners. Managing this powerful perceptual effect requires the competence of "critical listening," which Corey (2016) describes as "technical ear training" and Senior (2019) emphasizes as a holistic listening skill that distinguishes experts from novices. The presented formative framework serves as a bridge, translating this abstract skill of "critical listening" and the goal of "shaping tone" into concrete and actionable steps, such as adding "brightness" or "warmth." This is evidence of how abstract artistic intentions translate into their concrete and technical counterparts across the frequency spectrum, as demonstrated by Aras and Temuçin (2022).

The creative approach framed in this research represents the point at which the equalizer is used beyond problem-solving and aesthetic enhancement to radically transform sound as a sound design tool. The philosophical foundation of this approach lies in the concept of "Mixing as Performance," which views the mixing process as an intuitive and artistic act (Anthony, 2017; Izhaki, 2008). The presented creative framework offers a structure that combines this abstract, performance-based philosophy with concrete techniques found in the literature. For example, the "radio effect" scenario in the "Findings" section is a practical application of the "aging of audio files" technique described by Dutilleux et al. (2011). Therefore, the framework presented in this study defines the use of the creative equalizer as a teachable artistic process with specific goals and techniques, rather than a merely random and intuitive act.

Conclusion

This study analyzes the scattered literature on equalizer use in music production using a systematic review method and presents it within a holistic framework. The study determined that equalizer use serves three fundamental purposes: corrective, formative, and creative, and that these approaches can be structured within systematic frameworks. These findings reveal that equalizer use is not merely a technical troubleshooting tool; it is also a multilayered competency that shapes the aesthetic character, timbral value, and artistic expression of sound. The structured approaches and schematic visualizations presented in the study make this complex competency more understandable and teachable, filling an important gap in the literature, especially for audio engineering educators and novice users.

Recommendations

Based on the results of the study, the following recommendations were developed for practitioners, educators, and future researchers in the field:

Recommendations for Practitioners and Educators

Use of Structured Pedagogical Approaches: Structured training programs and practice guides based on the three fundamental approaches (corrective, formative, and creative) synthesized in this study should be developed in both institutions providing audio engineering education and for individual studies. Such structured frameworks can accelerate the learning process by making complex information more understandable.

Developing Critical Listening Skills: Instead of over-relying on the visual aids offered by digital tools when using equalizers, the development of critical listening skills—a core competency of sound engineers—should be encouraged. One of the key characteristics that distinguishes professional sound engineers from novices is the ability to translate perceptual impressions into sound technical decisions (Corey, 2016).

Contextual Adaptation and Creativity: Every musical production is unique in its own right. The approaches presented in this study should be viewed as a foundation, not a set of strict rules. Users should adapt and creatively utilize these frameworks based on their own artistic vision and the context of the project, as noted by master engineers (Owsinski, 2017).

Recommendations for Future Research

Analysis of Specific Approaches to Music Genres: Considering that different musical genres (classical, electronic, rock, jazz, etc.) have unique timbral goals and technical needs when using equalizers, more research should be conducted on genre-specific equalizer strategies.

Experimental Investigation of Psychoacoustic Effects: As noted in this study, timbre manipulations have tangible physiological and emotional effects on the listener. Experimental studies examining these effects of different equalizer manipulations will help us gain a deeper understanding of the tool's artistic potential.

Comparison of Human and Artificial Intelligence Interaction: Quantitative and qualitative studies should be conducted to compare the results of the human-centered approaches presented in this study with those of AI-powered automatic equalization systems. Such studies could contribute to the development of future hybrid systems by revealing the strengths and weaknesses of both approaches, as noted by Canyakan (2025).

Deepening Practice with Qualitative Research: Qualitative studies (interviews, case studies, etc.) examining the decision-making processes of music producers and tonmeisters in the recording, mixing and mastering stages and their philosophies regarding the use of equalizers will allow the introduction of experiential knowledge in this field to the academic literature.

Limitations of the Study

The findings and conclusions of this research should be evaluated within the framework of certain limitations. First, this study is inherently a theoretical synthesis, and the practical application of the presented approaches depends on many variables, including the hardware and software used, and the context of the production. The schematic drawings presented in the "Findings" section are idealized representations intended to illustrate fundamental principles and may not fully reflect the complexity of a real-world production.

Secondly, although the three approaches structured in this study constitute a systematic synthesis of the literature, their pedagogical effectiveness across different user groups has not been tested empirically. This remains an important topic for future research.

Finally, due to the nature of this review, the findings are limited by the scope of the 49 sources included in the analysis. The generalizability of the presented frameworks across different musical genres or production conditions requires further study.

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Artificial intelligence use transparency statement

The "Scispace" artificial intelligence software was used in the literature review phase of the research. Additionally, the "Jenni ai" software's paraphrase feature was used to analyze sentence structure suggestions for simplifying complex sentence structures. Direct quotes were not used.

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