# Digitalization of Sound Using Pulse Code Modulation (PCM)

**Siniša Tomić[1], Dalibor Drljača[2,3]**

[1]*Pan European University APEIRON Banja Luka, sinisa.m.tomic@apeiron-edu.eu*

[2]*Pan European University APEIRON Banja Luka, dalibor.p.drljaca@apeiron-edu.eu*

[3]*University Clinical Centre of Republic of Srpska, dalibor.drljaca@kc-bl.com*

**Case Study**

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**Abstract**: This paper focuses on Pulse Code Modulation (PCM) as a technology for digitizing analog signals. PCM is a widely used technique that enables precise encoding and transmission of analog information through digital pulse signals. The basic principles of PCM are explained here. PCM converts an analog signal into a digital form through sampling, quantization, and encoding. Sampling refers to the conversion of a continuous analog signal into discrete samples at regular time intervals. Then, quantization is applied to round each sample to the nearest possible quantization value, reducing the continuous range of values to discrete levels. Afterward, each quantized sample is encoded into a digital form. PCM is commonly used in various communication systems as well as in digital audio processing. In communication systems, PCM enables reliable transmission of voice signals, music, and other audio content over digital networks. In digital audio processing, PCM is used for recording, playback, and manipulation of sound, enabling high-quality reproduction and precise processing.

**Keywords:** PCM, Pulse Code Modulation, Analog Signal Digitization, Sampling, Quantization, Pulse Signals.

## Introduction

Sound is a time-varying mechanical deformation that propagates through a medium as a mechanical wave. [1, p. 1] The digitization of sound has brought numerous benefits in efficient storage, precise reproduction, sound manipulation, as well as compatibility and portability. This transformation has had a profound impact on the music industry, entertainment, communication, and our daily lives, enabling us to access, share, and enjoy sound in new and innovative ways.

Precise sound reproduction is a key advantage of digitization. Analog signals are prone to quality loss during transmission and reproduction due to various factors such as noise and interference. Digitization allows for accurate recreation of the original sound without any loss in quality, enabling compatibility and portability across different devices and systems, as well as integration with other media. Digital sound can easily be synchronized with video content, video games, or animations, creating cohesive multimedia experiences.

The availability of digital sound can vary depending on the context and specific situation. Having adequate technical infrastructure is a fundamental requirement for accessing digital sound. This includes broadband internet, mobile networks, computers, smartphones, digital players, and other devices that support the playback of digital audio formats.

Fast and reliable internet connection is crucial for streaming or downloading digital sound. High data transfer speed allows for uninterrupted streaming of audio content, while a stable connection minimizes playback interruptions.

Digital sound is available in various formats such as WAV, MP3, AAC, FLAC, and others. To utilize and play digital sound, support for the appropriate audio formats by devices, software, and applications is necessary.

Sound manipulation has become significantly more flexible thanks to digitization. Digital audio tools and software have enabled advanced sound processing techniques such as filtering, equalization, compression, and effects. This has opened the doors to creativity and innovation in music, film, and other areas of audio production.

## SUBJECTIVE CHARACTERISTICS OF SOUND

Subjective characteristics of sound are aspects of sound perception that arise in the human brain and depend on individual differences in hearing, psychological factors, and other influences. In the context of scientific research on sound, subjective characteristics are an important category that relates to the subjective experience of sound. Some of the most significant subjective characteristics of sound are loudness, pitch, sound colour, and deviations from the expected.

Loudness represents the subjective perception of sound intensity, which relates to the perception of the amplitude of the sound wave. Pitch is the subjective experience of the highness or lowness of a sound, which relates to the perception of the frequency of the sound wave. Sound colour is the subjective experience that pertains to the characteristics of sound that differentiate it from other sounds of the same frequency and loudness. Deviations from the expected are subjective experiences that occur when a sound has some unusual characteristic, such as noise, echo, interruption, frequency change, and others.

Subjective characteristics of sound play an important role in sound design, sound engineering, psychoacoustics, and other fields dealing with sound. Understanding the subjective characteristics of sound is crucial for the development of high-quality audio products such as headphones, speakers, microphones, and other sound reproduction technologies. Moreover, subjective characteristics of sound have an impact on creating emotional responses in listeners and influence how people experience sounds in everyday life. Therefore, a deeper understanding of the subjective characteristics of sound is a key element in the development and improvement of sound technology.

## OBJECTIVE CHARACTERISTICS OF SOUND

Objective characteristics of sound are physical quantities that can be measured and described based on the physical properties of sound waves.

Some of the most important objective characteristics of sound are:

- Frequency: the number of oscillations of a sound wave per unit of time (usually in seconds), expressed in Hertz (Hz).
- Amplitude: the maximum intensity of a sound wave, expressed in decibels (dB).
- Phase: the relationship between the phases of different sound waves that are interacting with each other.
- Wavelength: the distance between two points on a sound wave where the phase is the same, expressed in meters (m).
- Speed of sound: the rate at which sound propagates through a medium, expressed in meters per second (m/s).
- Spectrum: the set of frequency components that make up a sound.
- Duration: the length of time during which a sound can be heard, expressed in seconds (s).

These characteristics are fundamental elements of sound and represent key factors in its description and characterization.

## PCM - PULSE CODE MODULATION

Pulse Code Modulation (PCM) is a digital technique for encoding and transmitting analog signals. This technique is used for transmitting voice signals, music, and other analog signals in digital form. PCM enables the conversion of analog signals into a digital format, which can be transmitted through digital communication systems and smart devices.

PCM utilizes the processes of sampling and quantization to obtain a digital representation of the analog signal. First, the signal is taken at discrete intervals and digitized, which is known as sampling. Then, quantization is performed, which means that discrete levels of the signal are determined based on their values. These levels are called quantization levels, and the size of each discrete level is determined by the number of bits used to encode them.

Once the analog signal is quantized, coding is used to transmit the signal over the digital system. This is achieved by encoding each quantized sample as a digital code, which is then transmitted as a series of pulses. Pulse code modulation has the advantage of being easily applicable in digital data transmission systems.

The final output signal consists of a sequence of digital codes, representing the value of the signal at each sample. The size of the output signal depends on the number of bits used to encode each sample. A higher number of bits means higher precision but also increased memory requirements for storing the digital signal.

"In modern information systems, coding methods for transmitting and processing information effectively solve a large number of problems. PCM systems, like other systems involving quantization, exhibit a threshold effect. This means that external noise has no impact on the information processing processes as long as the signal-to-noise ratio remains below a certain threshold value. Once this value is exceeded, significant errors occur. Due to their robustness against interference and the ability to directly input digital information into electronic computers, PCM methods find extensive applications in modern information systems." [2]

PCM (Pulse Code Modulation) is applied in various digital systems, including telephony, CD players, digital audio and video formats, as well as in medicine for digital processing of medical images. Due to its efficiency, accuracy, and flexibility, PCM is considered a key technology in digital signal processing and telecommunications.

- *Filtering: removing unwanted frequencies from a signal, which improves signal quality and reduces noise.*
- *Sampling: the process of taking samples of an analog signal at regular time intervals to obtain a digital signal consisting of discrete values.*
- *Quantization: the process of converting a continuous analog signal into discrete values that can be represented digitally. Quantization usually refers to the quantization of signal amplitude.*
- *Encoding: the process of converting a digital signal into a form that can be transmitted over a digital communication channel.*
- *Regeneration: the process of restoring a signal after transmission over a communication channel to reduce the effect of losses and noise on the signal.*
- *Decoding: the process of converting an encoded digital signal back into the original digital signal.*
- *Filtering: the process of removing unwanted frequencies from a signal, improving signal quality and reducing noise.*
- *Transmitting side: the side that generates and sends the signal.*
- *Receiving side: the side that receives the signal and processes it.*

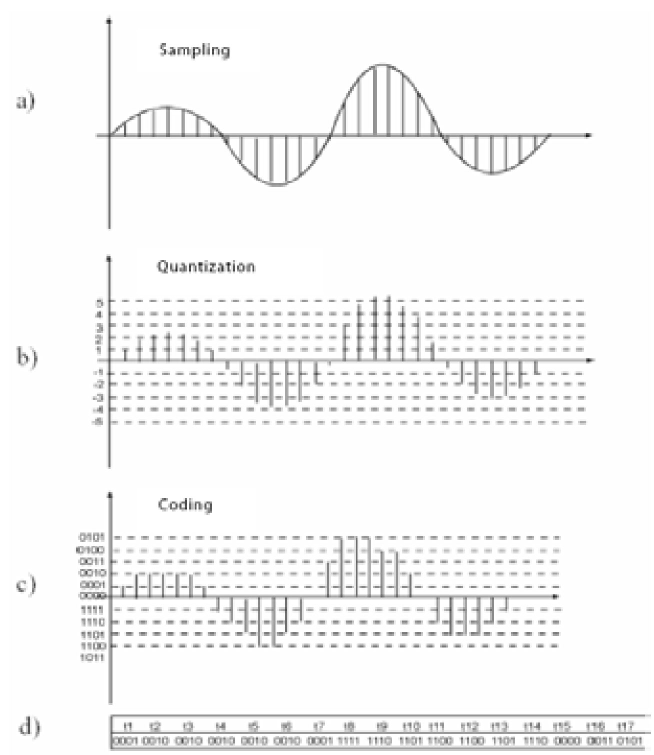


***Figure 2.*** *The process of PCM signal generation. [3, p. 10]*

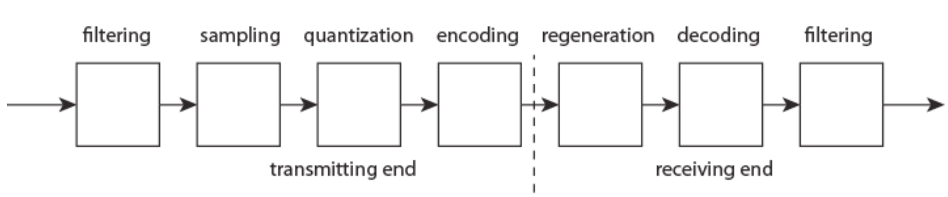


***Figure 1.*** *Block diagram of PCM signal generation and transmission. [3, p. 9]*

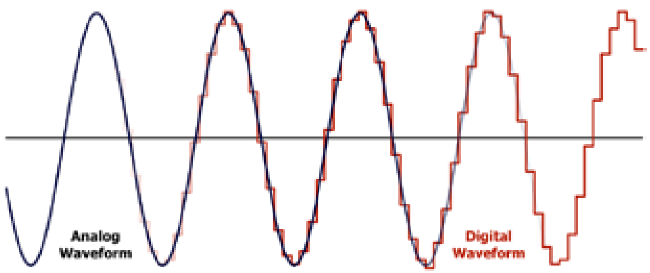


*Figure 3. An audio signal (black line) is sampled and quantized, with each sample assigned a digital value (red line) to create a digital approximation of the audio signal [4]*

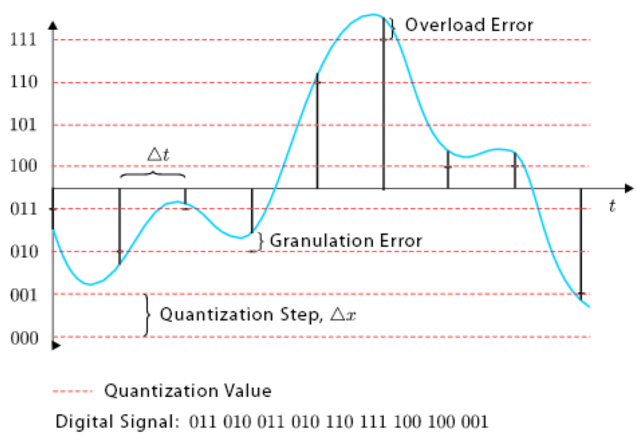


***Figure 4.*** *Graphical representation of the process of signal sampling and quantization. [5]*

## SAMPLING

Sampling is the process of converting a continuous analog signal into a discrete digital signal by taking samples at regular time intervals. Sampling is a crucial process in digital signal processing as it allows the analog signal to be transmitted and processed digitally using computer algorithms without significant loss of information from the original signal.

Sampling is typically performed at a specific frequency known as the sampling frequency. The sampling frequency needs to be sufficiently high to capture relevant portions of the analog signal while also being low enough to avoid unnecessary processing and storage costs. To fully reconstruct an analog audio signal into a digital audio signal, Nyquist's theorem needs to be applied.

Sampling greatly impacts the quality of the digital signal, so it is important to pay attention to the sampling process when designing digital signal processing systems and during audio recording and playback.

## NYQUIST'S THEOREM

Nyquist sampling theorem, also known as the Nyquist-Shannon theorem, was first published in 1928. [6, pp. 617-644] The theorem was formulated by American electrical engineer Harry Nyquist during his work at Bell Telephone Laboratories. Its mathematical support was later developed by Claude E. Shannon in 1949. The Nyquist-Shannon theorem provides a mathematical framework for sampling analog signals, transforming them into a discrete form that can be digitally processed.

It states that to accurately sample a signal, the sampling frequency (fs) must be at least twice the highest frequency (*fg*) present in the signal.

***fs = 2 x fg***

Since the human ear can perceive sounds up to 20 kHz, the minimum required sampling frequency would be 40 kHz. For added safety, compact discs have a slightly higher sampling frequency. [7]

***Sampling frequencies for various applications are as follows:***

*Telephones, wireless microphones: 8 kHz*
*G.722 VoIP: 16 kHz*
*Music CD: 44.1 kHz*
*Professional audio equipment: 48 kHz*
*Hi-Res Audio: Up to 192 kHz*

PCM was first widely used to digitize voice telephone communications to make it easier to transmit over long distances without any signal loss. The Digital Signal 0 (DS0) specification, originally set back in the 1960s and still used today, digitizes a phone call at 8 kHz at 8 bits per sample for a bitrate of 64 kbps. [4]

## QUANTIZATION

By quantization, we assign each sample value of the audio amplitude to the nearest predefined binary value, thus obtaining a digital representation of the sound.

A higher number of bits enables greater precision in quantization, resulting in better sound quality.

There are two types of quantization: uniform and non-uniform quantization. Uniform quantization uses equal spacing between quantization levels, which leads to distortion in sound due to quantization errors.

"The quantization error $e[k] = x[k] - y[k]$ occurs as a result of 'rounding' the signal level to a discrete value. It causes nonlinear distortion of the primary continuous signal $x(t)$, and it is a random variable expressed as quantization noise that appears alongside the useful signal. If the sample amplitude is close to the quantization interval boundary, the error can be largest. Since there is no predetermined relationship between quantization errors in neighboring intervals, the quantization error can be represented as a random signal with a uniform distribution" [8, p. 14]

Non-uniform quantization utilizes a non-linear quantization interval, reducing sound distortion and improving sound quality. The consequence of quantization is quantization noise, which manifests as sound distortion.

## QUANTIZATION NOISE

The consequence of quantization is quantization noise, which occurs when an analog audio signal is converted into digital form due to the discretization of the signal and rounding of sample values to the nearest quantization level. This discretization leads to an error in the value, and this error is perceived as background noise during the playback or recording of digital content, although in modern digital systems, quantization noise has a very low level and is rarely noticeable. Quantization noise arises when the analog signal is discretized and infinite signal values are replaced by a finite number of quantization levels. This noise can be reduced by using noise reduction methods such as dithering, which adds a small amount of noise to the audio signal before quantization to minimize distortion.

Clearly, coarser quantization results in more pronounced quantization noise, leading to poorer signal quality. To reduce quantization noise, finer quantization is required, meaning more bits per sample should be allocated. [1, p. 16]

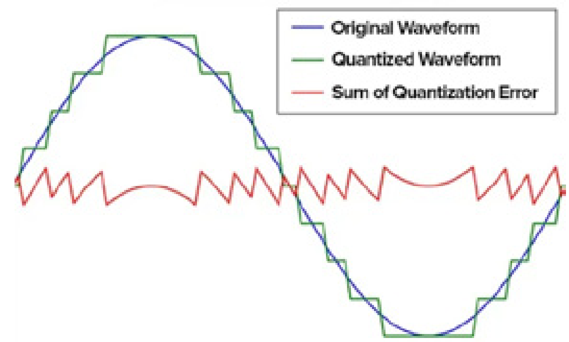


***Figure 5.*** *Quantization noise [9]*

## DITHERING

Dithering involves algorithms (coded directions given to a computer) that have to make decisions about how to handle the missing data (errors) from dropping out data in order to make the file sizes smaller.

That's the basic gist of the whole thing. Now, many websites will tell you that you're adding random noise that cancels out quantization error. That's not true. You're adding "noise" at specific places to try to smooth out gaps in data. This means that waveform analysis is occurring so particular choices can be made. There's nothing random about it. [9]

The purpose of dithering is to add a low-level broadband noise signal to the input signal, with a statistical energy distribution whose peak values correspond exactly to one quantization interval. [8, p. 16]

In other words, we add noise to reduce noise.

## ENCODING

Encoding involves assigning a binary code to each quantized sample. During sound reproduction, the process occurs in reverse order. The digital pulses are decoded into their original amplitude values, which are then filtered and converted back to analog form. The analog signal can be played through speakers to hear the original sound.

## CONCLUSION

PCM has become widely accepted and used in various industrial and communication applications. In communication systems, PCM enables reliable transmission of voice signals and audio content over digital networks, resulting in high sound quality and lossless transmission. In digital audio processing, PCM is used for recording, playback, and sound manipulation, providing high-fidelity reproduction and precise sound control.

The advantages of PCM include high accuracy and precision in transmission, data compression capability, and resistance to noise and interference. However, it requires sufficient bandwidth and data storage, and there are challenges in the quantization process that can affect signal quality.

PCM is a key technology for sound digitization, offering the ability to accurately encode, transmit, and process analog signals. Its applications encompass communication systems, digital audio processing, and other areas where the digitization of analog signals is crucial. Understanding and implementing PCM contribute to improving sound quality and broader access to digital audio in today's digital age.

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## ABOUT THE AUTHORS

**Siniša Tomić**, PhD, is an associate professor at the Pan-European University APEIRON, Banja Luka. He serves as the head of the study program "Computer Multimedia and Graphics". He actively participates in urban and international development projects, where he serves as an expert lecturer. He is the author of multiple scientific and professional papers published in reputable scientific journals and conference proceedings. Areas of expertise: audio engineering, computer graphics, 2D and 3D computer animation, digital special effects, digital video production and post-production, virtual space creation, advanced studio techniques, software design, digital print, digital photography and videography, e-learning, education, development, ...

**Dalibor P. Drljača**, PhD, is an associated expert in Research Centre of the University Clinical Centre of Republic of Srpska. He is also engaged as an assistant professor at the Pan-European University APEIRON in Banja Luka in the teaching process. His background is in information systems and public administration, but also has a significant experience with European Union funded projects in research and innovation. Last couple of years was engaged by the European Commission as a expert and trainer for EU organised events in South East Asia and Africa. Has published more than 30 papers and published six books.


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