http://doi.org/10.35784/iapgos.6118

received: 20.04.2024 | revised: 27.05.2024 | accepted: 22.06.2024 | available online: 30.06.2024

MEANS OF ANALYZING PARAMETERS OF SPEECH SIGNAL TRANSMISSION AND REPRODUCTION

Oleksiy Azarov[1], Larysa Azarova[1], Iurii Krak[2,3], Leonid Krupelnitskyi[1], Angzhelika Azarova[1], Veronika Azarova[4]

[1]Vinnytsia National Technical University, Vinnytsia, Ukraine, [2]Taras Shevchenko National University of Kyiv, Kyiv, Ukraine, [3]Glushkov Cybernetics Institute, Kyiv, Ukraine, [4]Borys Grinchenko Kyiv University, Kyiv, Ukraine

Abstract. *The article explores the design and use of the analyzer developed by the authors for investigating the quality of compression, recording, transmission, and reproduction of speech signals in audio communication channels. The peculiarities of the construction of measurement channels of a self-adjusting system are studied. Parameters for assessing the quality of channels are selected, and methods for their calculation are determined. An example of determining the quality of compression-decompression in satellite channels for transmitting audio speech is provided. A polyharmonic signal, consisting of the sum of many harmonic components with the same amplitude and phase, is proposed as the input test signal for the compressor. Such a signal approximates real speech and music signals transmitted through the channel. The actual frequency characteristic is determined at the output of the decompressor by analyzing the spectrum, which, for the MPEG Layer3 compression method, significantly depends on the bit rate of compressed data transmission. Another characteristic application of the analyzer is experiments to study the structural organization of connecting possibilities of elements in complex nominations at the phonetic level in the concept of the "golden" ratio. A hypothesis is put forward about the harmony of these words at the acoustic level. It is proved that the harmonious structure of juxtapositions should be considered as the arrangement of components of such nominations by a system of perfect relationships, among which the proportion corresponding to the division of the whole into two parts and called the "golden" ratio plays a fundamental role.*

Keywords: sound signal analysis, analog-to-digital conversion, sound signal compression, acoustic phonetics

SPOSOBY ANALIZY PARAMETRÓW TRANSMISJI I ODTWORZENIA SYGNAŁU MOWY

Streszczenie. *W artykule omówiono budowę i zastosowanie opracowanego przez autorów analizatora do badania jakości kompresji, rejestracji, transmisji i odtwarzania sygnałów mowy w kanałach komunikacji audio. Badano specyfikę budowy kanałów pomiarowych układu samoregulacji. Wybrano parametry do oceny jakości kanałów i określono metody ich obliczania. Podano przykład określenia jakości kompresji-dekompresji w kanałach satelitarnych do transmisji mowy. Jako wejściowy sygnał testowy dla kompresji zaproponowano sygnał harmoniczny, składający się z sumy wielu składowych harmonicznych o tej samej amplitudzie i fazie. Sygnał taki jest zbliżony do rzeczywistych sygnałów mowy i muzyki przesyłanych kanałem. Rzeczywista charakterystyka częstotliwościowa jest określana na wyjściu po dekompresji poprzez analizę widma, które w przypadku metody kompresji MPEG Layer3 w istotny sposób zależy od szybkości transmisji skompresowanych danych. Kolejnym charakterystycznym zastosowaniem analizatora są eksperymenty mające na celu badanie strukturalnej organizacji możliwości łączenia elementów w złożonych zespołach na poziomie fonetycznym w koncepcji „złotego" podziału. Postawiono hipotezę o harmonii tych słów na poziomie akustycznym. Udowodniono, że za harmonijną strukturę zestawień należy rozumieć uporządkowanie składników poprzez system doskonałych relacji, wśród których istotną rolę odgrywa proporcja odpowiadająca podziałowi całości na dwie części, zwana „złotym" podziałem.*

Słowa kluczowe: analiza sygnału dźwiękowego, konwersja sygnału analogowo-cyfrowego, kompresja sygnału dźwiękowego, fonetyka akustyczna

Introduction and literature review

In the era of global computerization, where language serves a communicative function, the use of information technologies in linguistics becomes increasingly relevant for both theoretical research and solving applied tasks. This allows for a deeper exploration of various linguistic phenomena through the analysis of language functioning and human language activity processes since language is not only a means of conveying thoughts but also a way of representing knowledge.

Computational linguistics is a branch of linguistics that studies language using information technologies and creates linguistic support for information processing computer systems. The emergence of computational linguistics is driven by society's need for new operational methods of processing linguistic information. A broad understanding of the subject of computational linguistics has developed in global science, addressing both fundamental theoretical and applied tasks of modern linguistics [13].

In addition to basic classical principles, innovative forms associated with the use of state-of-the-art information technologies, measurement systems, multimedia software, computer-based language learning programs, etc., are gaining importance today. The integration of computer technologies requires further research and improvements, as it is essential to incorporate informational tools into linguistics that allow scholars to study language systems at a modern scientific level [6].

In the study of oral language, the following levels are distinguished:

1) physiological level of language formation;
2) articulation level, including observation, arrangement, and forms of speech organs, determining the dimensions of acoustic cavities, etc.;
3) acoustic level, where physical sounds are analysed using equipment;
4) level of ear physiology and nerve processes, physiologically related to hearing;
5) psychological level related to the problem of word recognition in the stream of sounds in complex environmental conditions [9]. The first and fourth levels are the subject of research by linguists-physiologists, and the fifth is studied by psycholinguists.

Well-known attempts at computerizing the research on the quality of speech usually rely on approaches defined by standards in the field of telecommunications and broadcasting. In a comprehensive study [10], the quality of voice and audio signals was evaluated based on the recommendations of the International Telecommunication Union (ITU). Evaluation methods used recommendations P.800 (MOS scale), P.862 (PESQ), P.862.2 (W-PESQ), and P.563 for speech evaluation and BS.1116, BS.1534, and BS.1387 (PEAQ) for the overall assessment of sound signals. This work experimentally illustrates the comparison of objective and subjective methods of speech quality evaluation using the mentioned methodologies. However, the measurement tools used and their usage schemes are not specified, reducing the repeatability of similar experiment results.

In the work [12], the mentioned objective methods for evaluating speech quality are utilized, and the authors of this article propose a polyharmonic method for investigating frequency characteristics, as presented in [3]. This polyharmonic method is intended for developing software to determine specific parameters of the quality of analog and digital sound equipment tracts. The measurements use a computer's sound card, which lacks standardized metrological characteristics, and methods of self-calibration are not employed.

The issue of adequately assessing the quality of speech transmission and perception sharply arises in the design and use of hearing aids for speech perception [8] and music [11], regenerating gaps in speech in text-to-speech synthesis systems [15], etc. These studies typically use expert (subjective) methods for quality evaluation, although they may apply statistical methods for processing their results.

Such an approach does not ensure experiment repeatability in different conditions and may lead to ambiguity in decisions based on quality evaluation results.

Therefore, considering the variety of required measurements for evaluating the quality of speech and the diversity of the tracts themselves, we conclude on the relevance of creating a comprehensive high-precision measurement system for determining unified, standardized parameters of sound tracts.

The subject of the study is the methods of construction and application of specialized computer measurement systems in the objective evaluation of the quality parameters of sound channels and tracts, as well as in the analysis of speech signal parameters.

1. The aim and objectives of the study

The aim of the article is to increase the accuracy and reliability of measurement results when studying the quality parameters of compression, recording, transmission, and reproduction of signals in electronic sound channels, particularly in evaluating the quality of speech compression signals and studying the structural organization of complex words at the phonetic (acoustic) level.

Achieving this goal involves performing the following problems:

- Determine the technical features of the construction and operation of hardware and software for measurements in quality evaluation.
- Present the ranges and metrological characteristics achieved in the specialized computer system for analysing parameters of sound tracts.
- Propose a method for determining the real amplitude-frequency characteristic during the transmission of sound signals through compressed channels.
- Investigate the structural organization of connecting possibilities of elements in complex nominations at the phonetic level in the concept of the "golden" ratio.

2. Development of specialized computer systems for objective measurement of sound channels and speech processing tracts

To achieve the set goal and perform the research tasks, the authors of the article propose an approach that involves the development and use of specialized analog-to-digital computer systems for the objective measurement of sound transmission channels and speech signal processing tracts. These systems should comply with current standards for sound transmission quality, including ITU-R BS.644-1 recommendations and the corresponding Ukrainian standard [14].

A series of studies on the development and implementation of high-precision self-correcting analog-to-digital systems were conducted by the authors with a group of developers from Vinnytsia National Technical University [1, 2]. These systems notably use weight redundancy based on Fibonacci codes and the "golden" ratio in analog-to-digital and digital-to-analog converters (ADC and DAC). Correction methods are also applied to the measurement channels of the systems, characterized by the use of modern methods of current amplification and signal switching [4, 5].

For the tasks of phonetic analysis of speech signals, a specialized measurement and information system called the Sound Tract Parameters Analyzer (STPA) has been developed. The system is designed for automated measurement of electrical characteristics and parameters of tracts for the formation, processing, and transmission of radio programs and television audio, low-frequency tracts of radio electronic devices, tape recorders, level indicators, etc.

The STPA uses an original measurement scheme (figure 1): the computer calculates the measurement signal, sends it in digital form to the DAC, then the signal in analog form enters the input of the investigated sound tract, and from its output – to the input of the ADC of the analyzer and returns to the computer, but by this time, it already has distortions introduced by the sound tract. Analyzing these distortions with the STPA program allows for a complete determination of the quality of the sound tract. The generator and STPA meter also have separate operating modes, measurements of channel and tract parameters distributed in space (intercity and satellite transmission channels, cable and relay lines), and in time (recorded on tape, optical, or other media).

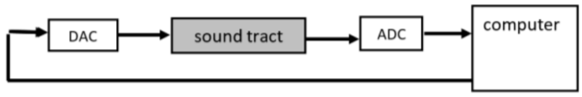


Fig. 1. Basic Measurement Scheme Using STPA

The versatility of this approach allows replacing over 15 "classical" measuring instruments with the STPA when measuring sound path parameters: low-frequency generator, root mean square voltmeter, selective voltmeter, frequency meter, phase meter, non-linear distortion and difference tone meter, spectrum analyser, amplitude-frequency response (AFR) and group delay time (GDT) meters, integral and psychometric noises meters, interchannel interference, parasitic amplitude modulation, signal-to-noise ratio, etc.

The structural diagram of the STPA system with the designation of its main components and the path of the measurement signal is provided in figure 2.

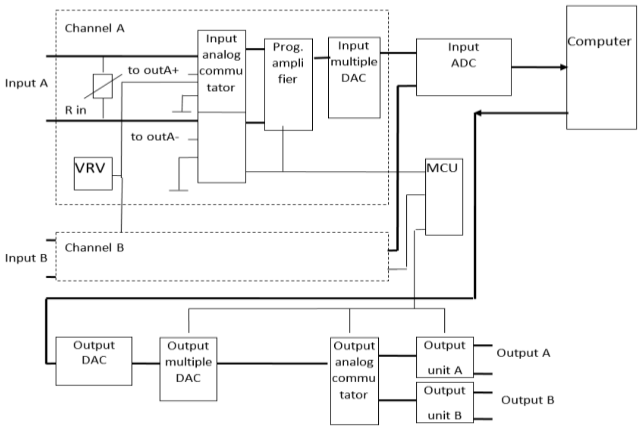


Fig. 2. Structural Diagram of STPA

The input section consists of two measurement channels, A and B. The differential inputs have programmable resistors to adapt to different sound tracts. The analog input switch has the main signal input, as well as inputs for the signal from the DAC, the source of variable reference voltage (VRV), and the "zero" bus (used for calibration of gain coefficients and self-tuning of the frequency and phase-frequency characteristics (AFR and GDT) of the two measurement channels A and B. The programmable amplifier adjusts one of the 4 ranges of input voltage from minus 24 dB to +12 dB (0 dB – 0.775 V), and the input DAC with multiplication adjusts the level of the ADC input voltage with a step of 0.1 dB from minus 30 dB to +18 dB. The ADC converts analog signals with an accuracy of 20...24 binary bits and a sampling frequency of 44.1...96 kHz. The output code of the ADC via the USB interface goes to the personal computer, where digital signal processing occurs according to the measurement algorithm. Control of the analog part is carried out from the computer through the microprocessor control unit (MCU).

The STPA generator contains the DAC of the main signal and the output multiplication DAC. By sequentially connecting these DACs, it is possible to smoothly change the voltage level of the output signal from minus 80 to +12 dB with a step of 0.1 dB. The analog output switch and output devices A and B consist of buffer differential amplifiers. Structurally, the measurement part of the STPA is made in the form of a small module with dimensions of 297x210x40 mm.

The main method of signal processing in the STPA is Fast Fourier Transform (FFT), which allows obtaining spectra of signals that have passed through sound channels and tracts. Any distortions of the shape of the reference signal created by the DAC are determined by its spectrum. To achieve sufficient frequency resolution, FFT is used for 16384 points, corresponding to a sampling duration of 0.372 s and a resolution of 2.69 Hz at a sampling frequency of 44.1 kHz. Input signals with frequencies not multiples of the specified frequency lead to the so-called "blurring" of the spectrum between adjacent harmonics. To reduce "blurring," the input signal sample before FFT is multiplied by a window weight function that has slopes at the edges, reducing edge effects. Hanning and Blackman-Harris windows are used.

The Hanning window, used in the analysis of nonlinear distortions, AFR, and GDT, is defined as follows:

$$W(i) = 0.5 - 0.5\cos\left(2\pi\frac{i}{n}\right), i = 0 \dots n-1$$

where n is the number of FFT points.

The Blackman-Harris window, which is employed in noise and selective interference analysis, has more components:

$$W(i) = 0.35875 - 0.48829\cos\left(2\pi\frac{i}{n}\right) + 0.14128\cos(4\pi\frac{i}{n}) - 0.01168\cos(6\pi\frac{i}{n}), i = 0 \dots n-1$$

If a harmonic test signal is generated and reproduced through the DAC:

$$S(i) = A_0\sin\left(2\pi f_0\frac{i}{n}\right), i = 0 \dots n-1$$

and passed through the measured sound tract, the output signal's spectrum will contain harmonics with frequencies 2 f_0, 3 f_0 and so on, in addition to the fundamental frequency f_0. The nonlinear distortion coefficient is then calculated as the ratio of the root mean square sum of the first m = 3 or m = 5 distortion components to the amplitude of the fundamental signal:

$$K = \frac{\sqrt{\sum_{j=2}^{m} A^2(jf_0)}}{A(f_0)}$$

Additionally, the phase difference between channels A and B of the stereo pair can be determined using the same signal since the signal sample is taken simultaneously on both channels.

For the measurement of AFR, GDT, and phase response, a polyharmonic test signal is used [3]:

$$S(i) = \sum_{j=1}^{k}(A_J\sin(2\pi f_j\frac{i}{n}), i = 0 \dots n-1$$

where k is the number of simultaneously specified harmonics.

In the output spectrum of such a signal after passing through the measured sound tract, the amplitudes of the harmonics are proportional to the AFR of the tract. Simultaneously, the phase response is determined.

Measurement of integral (unweighted) and psychometric (weighted) noise is carried out in the time domain by determining the quasi-peak value of the signal within an interval of up to 10 s. Weighting involves digital filtering with a psychometric characteristic approximating the sensitivity function of the human ear.

Methods for measuring other properties of sound channels are also based on digital signal processing, replicating the functions of respective analog measuring devices in standardized measurement schemes. For more complex measurement methods (such as frequency, detonation, ultrasonic characteristics, etc.), algorithms have been developed that have also undergone comprehensive metrological certification according to methodologies adopted for measuring and information systems. To achieve high measurement accuracy, the analyser includes automatic calibration and self-correction of measurement results, taking into account its own systematic errors. Self-correction involves adjusting the transfer coefficients, frequency response, and phase response of the measuring channel.

With the number of measured values and the measurement ranges (table 1), the developed Sound Tract Parameters Analyser STPA fully complies with the requirements of Recommendation ITU-R BS.644-1:1990, IDT, and the corresponding standard of Ukraine [14].

Table 1. Main parameters of sound channel sand zones that can be determined using the STPA and their measurement ranges

Parameters	Range of Values
Root Mean Squareand Selective Voltage	+20 … -90 dB
Integral Tapsofometric Noises	-20 … -90 dB
Level of Selective Interference	-20 … -90 dB
Frequency of input signals	20 … 20000 Hz
Bandwidthof Frequency Response (BFR)	20 … 20000 Hz
Bandwidth of Phase Response, Phase Difference	-90 … +90 degrees
Group Delay Time	0.5 … 50 ms
Nonlinear Distortion Coefficient	0.05 … 5%
Difference Tone Coefficient	0.05 … 5%
Intermodulation Coefficient	0.05 … 5%
Detonation Coefficient	0.01… 5%
Signal Duration for Level Indicator Checking	1 … 220 ms
Transient Interference Level	30 … 80 dB
Level of Parasitic AmplitudeModulation	30 … 80 dB

3. Experiments

3.1. Application of STPA in evaluating the quality of digital sound channels with compression

The application areas of STPA in the authors' research are diverse, covering complex and current tasks related to the evaluation of low-frequency audio and speech signals, such as sound fragment recognition and acoustic surface reconstruction, among others. The methods for analysing and processing measured acoustic signals are based on both deterministic and neural-fuzzy technologies [4, 5].

Let's consider a typical example of STPA application – evaluating the quality of channels and tracts of digital sound transmission with compression. To measure the AFR and GDT of sound transmission channels, a polyharmonic signal is used [3], composed of the sum of 10 to 50 harmonic components, whose frequencies are selected from a standardized series or arbitrarily. An example of the oscillogram of a polyharmonic signal is shown in figure 3, and its spectrum in figure 4 (using the standardized frequency range of 32.5, 63, 125, 250, 500, 1000, 2000, 4000, 10000, 16000 Hz).

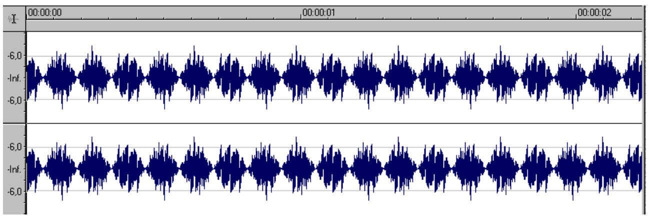


Fig. 3. Shape of the polyharmonic test signal (10 frequencies)

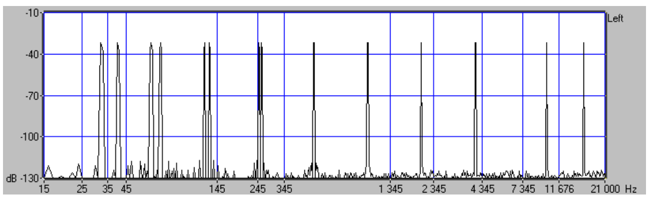


Fig. 4. Spectrum of the polyharmonic test signal (10 frequencies)

The method of measuring AFR using a polyharmonic signal provides significant advantages in managing tracts that contain sound compression and decompression algorithms, such as MPEG. Compression can reduce the required symbol transmission rate and channel bandwidth but may decrease signal quality. Traditional methods, using monoharmonic or frequency-modulated signals for AFR determination, do not show deviations from the norm. However, subjective listening by experts of real speech programs through these channels often yields an unsatisfactory assessment of AFR compliance with the declared quality level.

A polyharmonic signal consists of many frequency components, making it more similar to real sound signals. For this reason, the proposed polyharmonic frequency response method more objectively assesses the quality of sound signal transmission through channels and voice tracts. Figure 5 shows the spectrum of the polyharmonic signal after transmission through a typical satellite channel with a bandwidth of 128 kbit/s and an appropriate MPEGLayer3 compressor/decompressor.

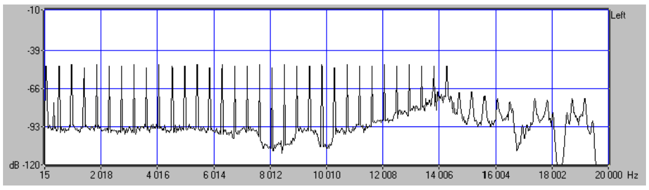


Fig. 5. Spectrum of the polyharmonic signal after passage through a channel with MPEGLayer3 compression 128 kbit/s (45 frequencies)

Several compression methods were tested with different channel bandwidths. It is worth noting that the best results were achieved with the MPEGLayer3 compression method with a bandwidth of 384 Kbit/s, the spectrum of which is shown in Fig. 6.

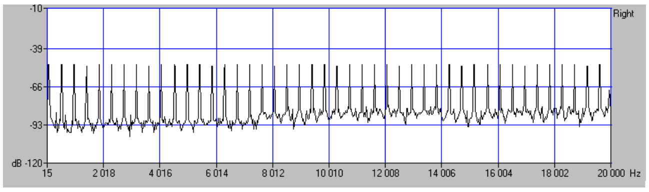


Fig. 6. Spectrum of the polyharmonic signal after passage through a channel with MPEGLayer3 compression 384 Kbit/s (45 frequencies)

3.2. Example of APZT application in studying noun-yuxtaposits at the phonetic level in the concept of the "Golden Ratio"

Let's delve into another characteristic example of STPA application in studying noun-yuxtaposits at the phonetic level in the concept of the "golden ratio".

When comparing oral and written speech, it can be said that oral speech is characterized by sophistication, conciseness, and speed. Through intonation, changes in tempo, variations in voice pitch, and articulation, oral expression can be endowed with greater expressiveness and flexibility. It's noteworthy that the results of oral language research interest not only linguists but also psychologists, physiologists, communication engineers (acousticians), etc. This is associated with the use of hardware as an information-measuring system for speech signals.

A linguist-phonetician is primarily interested in the physical, anatomical-physiological, and linguistic aspects. The study of the physical or acoustic nature of speech sound involves determining its pitch, intensity, tonality, timbre, and temporal duration. The study of speech sound from an anatomical-physiological point of view is motivated by the fact that it is created by the speech organs of a person. The essence of this analysis lies in determining the nature and degree of involvement of various speech organs in the process of sound formation.

The third aspect of studying human speech sounds is linguistic or phonological. The most important task of the linguistic aspect is to analyse sounds as functional units of the language system. The study of speech sounds in the phonological aspect is considered crucial in linguistics, while its acoustic and anatomical-physiological study plays an auxiliary role. Oral expression has its musical form, rhythm, melody, and harmony. Melody is a specific sequence of sounds in time, and harmony is a set of sounds present simultaneously. Similarly, language is a stream of sounds, which is divided into temporal sequences of phonemes. It can also be represented as a series of coordinated actions controlling the vocal tract, larynx, lips, tongue, and teeth. In this case, linguistic transcription is similar to musical notation. Phonemes correspond to a sequence of chords, forming not a single melodic line but a fourfold, sixfold, or even more complex harmony [9]. Of course, this is only an approximate analogy, but a phoneme can be considered as a chord or a combination of sound features in a chord.

Phonetic analysis revealed the "golden" ratio in yuxtaposits by assessing the ratios of temporal intervals. The measurement of these time intervals was carried out using specialized digital sound processing equipment. The lengths of complex words were considered as time intervals in the process of sounding. The durations of temporal intervals assigned to these segments were estimated. A millisecond (ms) was chosen as the unit of measurement. The use of sound recording and playback devices allowed for a clear determination of the division center of a complex unit into components, measuring the duration of a whole word and its parts, and storing this information in computer memory.

Proportion indicators are found from the ratio [6]:

$$P_f = \frac{\text{Duration of the whole word (ms)}}{\text{Duration of the major component (ms)}}$$

where the symbol P_f defines the proportion level at the phonetic level.

We propose the following sequence for studying the proportions of noun-yuxtaposits construction. Firstly, we systematize the identified complex nominations by origin and part-of-speech membership. The second stage involves the process of "voicing" these nominations. The third stage involves temporal research of units using intonograms, which allow evaluating the ratio of the construction of a whole word and its components. In the final stage, the calculated value of P_f is compared with the data in the tables for "golden" p- and s-proportions. Based on the results of the analysis, the words are classified into the respective families [7].

It is worth noting that by analysing the numerical indicators in the ratio of sound durations of components in noun-yuxtaposits at the acoustic level, we can observe certain regularities in their construction. Specifically, their quantitative indicators are related to the "golden" proportions *p* and *s*. "Golden" proportions manifest themselves in various natural phenomena, such as human body proportions, plant structures, phyllotaxis, atomic compounds in molecules, brain function, mineral crystals, nuclear reactions, etc. The discovery of the "golden" ratio in various natural phenomena provides grounds to believe that there are universal criteria for harmony both in nature and in human creativity, particularly in poetry, architecture, and art. Additionally, there are mathematical equations whose solutions yield "golden" or harmonic proportions [6, 7]. The values of "golden" p-proportions are calculated as the real positive roots of the polynomial

$$x^{p+1} - x^p - 1 = 0$$

The solutions of this polynomial provide the values α_p, as shown in table 2.

Table 2. Values of p-proportions

p	0	1	2	3	4	5	…	∞
α_p	2.0	1.618	1.465	1.38	1.324	1.285	…	1.0

For "golden" s-proportions, the values $\alpha_s$are found as the real positive roots of the polynomial

$$x^{S+1} - \sum_{i=0}^{S} x^i = 0$$

and the solutions are presented in table 3.

Table 3. Values of s-proportions

s	0	1	2	3	4	5	∞
α_s	1	1.618	1.839	1.928	1.966	1.98	2.0

It is noteworthy that in the range of values between the classical "golden" proportion ($\alpha_{p=1}$ ≈1.618) and the s-proportion ($\alpha_{S=2}$ ≈1.839), there is a set of intermediate numbers that can also be computed as solutions of the polynomial

$$x^{l+2} - x^{l+1} - x^l - 1 = 0$$

The values of these solutions for certain l are listed in table 4.

Table 4. Values ofl-proportions

l	1	2	3	4	5	6	7
α_l	1.839	1.755	1.705	1.674	1.654	1.641	1.632

Let's consider the proportion of the construction of the yuxtaposit "chlen-korespondent" (member-correspondent): P_f = (417 ms + 1015 ms)/1015 ms = 1.41.

Similarly, for the yuxtaposit "boiar-zillia" (boyar-herb): P_f = (501 ms + 773 ms)/773 ms = 1.65. The duration of their sound is indicated above the components in milliseconds. By comparing these results with the data in the table of fractional indicators of "golden" p-proportions, it can be asserted that the nominative unit on Ukrainian "chlen-korespondent" belongs to the "golden" 3-proportion family, while the yuxtaposit "boiar-zillia" belongs to the "golden" 1-proportion family.

The intonograms of the words "chlen-korespondent" (member-correspondent) and "boyar-zillia" (boyar-herb) are illustrated in figures 7 and 8, respectively, with corresponding phonetic durations.

Chlen-korespondent:
Duration: 417 ms + 1051 ms
Phonetic Ratio: P_f = (417 ms + 1051 ms)/1051 ms ≈ 1.41
Boiar-zillia (boyar-herb):
Duration: 501 ms + 773 ms
Phonetic Ratio P_f = (501 ms + 773 ms)/773 ms ≈ 1.65

Phonetic Analysis of Other Yuxtaposits:
Yunak-lytsar (Youth-knight):
Duration: 453 ms + 635 ms
Phonetic Ratio P_f = 1088 ms / 635 ms ≈ 1.71

Liuks-sekunda (Lux-second):
Duration: 530 ms + 801 ms
Phonetic Ratio P_f = 1331 ms / 801 ms ≈ 1.66
Roztiah-stysk (Stretch-compression):
Duration: 1126 ms + 697 ms
Phonetic Ratio P_f = 1823 ms / 1126 ms ≈ 1.619

Tsar-zillia (Tsar-herb):
Duration: 696 ms + 987 ms
Phonetic Ratio P_f = 1683 ms / 987 ms ≈ 1.705

Holka-ryba (Needle-fish):
Duration: 444 ms + 525 ms
Phonetic Ratio P_f = 969 ms / 525 ms ≈ 1.85

Khlib-sil (Bread-salt):
Duration: 383 ms + 553 ms
Phonetic Ratio P_f = 936 ms / 553 ms ≈ 1.69

Fig. 7. Intonogram of the word "chlen-korespondent" (member-correspondent)

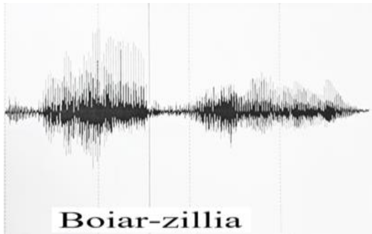


Fig. 8. Intonogram of the word "Boiar-zillia" (boyar'sherb)

4. Results and discussions

The intonograms of the examined words indicate that the majority (35 yuxtaposits) belong to the "golden" 1-proportion family, 8 words belong to the "golden" 2-proportion family, and 3 words belong to the "golden" 3-proportion family. There is one word each for the "golden" 4-proportion and "golden" 5-proportion families.

A notable aspect is the analysis of the information transfer process from the speaker to the listener. However, uncertainties may arise, including:

Uncertainty of speech sounds or acoustic images.

Uncertainty in language and syntax – sentence structure may vary.

Spoken language may deviate from syntax rules.

Ambiguity due to external conditions, such as noise, phone calls, and other conversations.

Recognition uncertainty, which depends on individual speaker characteristics and familiarity with language skills.

Despite these uncertainties, successful language communication often occurs. Overcoming the harmful effects of uncertainties is facilitated by the unique structure of language, which exhibits linguistic redundancy at various levels: phonemes, syllables, lexemes, sentences, and, ultimately, the entire text.

The problem of reducing linguistic redundancy is highly relevant and requires a higher level of information protection. Linguists and experts in automated text processing actively work on solutions. During automated text processing, the Zipf-Mandelbrot laws and statistical distribution laws of linguistic units are often utilized. Attention is focused on evaluating the required redundancy magnitude. Shannon's information theory, developed in the late 1940s and early 1950s, plays a significant role in this regard. However, applying such approaches to human language is challenging, as linguists need more tangible indicators than "redundancy associated with alphabet entropy" borrowed from information theory. Linguistic redundancy is also evident in the construction of complex nominations. The research on yuxtaposits achieved only a 1.0% error rate, demonstrating that "golden" proportions and their invariants are widely represented in the construction of complex words, serving as quantitative indicators of harmony. Natural language filtration eliminates awkward, unsuccessful, and non-harmonious constructions in complex nominations.

5. Conclusions

The utilization of computer methods for analysing the parameters of transmission and reproduction of linguistic signals allows for the effective assessment of the quality of audio channels and prompt determination and adjustment of their technical characteristics. For instance, in the analysis of the quality of satellite channels for transmitting audio signals with compression using the MPEGLayer3 algorithm, it was revealed that the use of a special polyharmonic signal enables the detection of significant distortions in the frequency response. These distortions can be rectified by selecting a higher bandwidth for the digital data transmission channel. In another example, during the phonetic investigation of complex nominations, it was found that noun juxtapositions exhibit a harmonious structure. This is corroborated by the values of proportions of sound durations for the entire complex word and its major component.

Research into the harmonious construction of noun juxtapositions at the acoustic level indicates that dominant units adhere to the classical "golden" 1-proportion, while others correspond to the quantitative indicators of "golden" p and s proportions. All of this attests to the non-random presence of clearly defined temporal relationships at the phonetic (acoustic) level, suggesting a harmonious structure in complex nominations. This is attributed to the influence of the human brain on the process of language formation in general and on complex nominations in particular. In the study of noun juxtapositions at the phonetic level within the framework of the "golden" proportion concept, a linguo-informational approach was employed, allowing the combination of linguistic methods and techniques with various positions of information theory as an information-measuring system for linguistic signals.

The prospects for further development of the discussed problem involve the use of computer measuring devices to evaluate the quality of transmitting audio language signals in digital compression systems in the presence of analog obstacles and digital errors in transmission channels and playback paths.

References

[1] Azarov A. D. et al.: Selection of the calculus system base for ADC and DAC with weight redundancy. Proc. SPIE 11176, 2019, 1117662 [https://doi.org/10.1117/12.2537197].

[2] Azarov O. D. et al.: AD systems for processing of low frequency signals based on selfcalibrate ADC and DAC with weight redundancy. Przegląd Elektrotechniczny 93(5), 2017, 125–128 [https://doi.org/10.15199/48.2017.05.26].

[3] Azarov O. D. et al.: Polyharmonic methods for measuring frequency characteristics of audio channels and tracts. Information Technologies and Computer Engineering 2, 2015, 23–29.

[4] Azarov O. et al.: Reconstruction of Acoustic Surfaces from Incomplete Data as an Identification Problem Based on Fuzzy Relations. CEUR Workshop Proceedings 3101, 2021, 208–226.

[5] Azarov O., Krupelnitskyi L., Rakytyanska H.: Sound field reconstruction from incomplete data by solving fuzzy relational equations. Advances in Intelligent Systems and Computing 1246 AISC, 2021, 547–566 [https://doi.org/10.15199/48.2017.05.2610.1007/978-3-030-54215-3_35].

[6] Azarova L. Ye.: Features of using computer technologies in the study of noun juxtapositions at the phonetic level. Linguistic Studies of Donetsk National University 36, 2018, 136–141.

[7] Azarova L. Ye.: Research of two-component complex nomination sat the phonetic level in the concept of the "golden" ratio. Scientific notes of Taurida National V. I. Vernadsky University. Series: Philology. Journalism 33(72), 6, Part 1, 2022, 1–6.

[8] Bispo B., Borges R.: A Cepstral Method to Estimate the Stable Optimal Solution for Feedforward Occlusion Cancellation in Hearing Aids. Journal of Communication and Information Systems 35, 2020, 113–123 [https://doi.org/10.14209/jcis.2020.12].

[9] Cherry C.: On Human Communication. A Review, a Survey, and a Criticism. Cambridge (Mass.). John Wiley & Sons, New York, London, 1972.

[10] de Lima A. A. et al.: On the quality assessment of sound signals. Proc. IEEE International Symposium on Circuits and Systems, 2008, 416–419 [https://doi.org/10.1109/ISCAS.2008.4541443].

[11] Kates J., Arehart K.: The hearing-aid audio quality index (HAAQI). IEEE/ACM Transactions on Audio, Speech, and Language Processing 24, 2015 [https://doi.org/10.1109/TASLP.2015.2507858].

[12] Kyryichuk D. L., Bolotin O. A., Liashenko O. M.: Development of a software for testing the quality of analog and digital paths of sound equipment. Scientific notes of Taurida National V. I. Vernadsky University. Series: Technical Sciences 5, 2022, 100–106 [https://doi.org/10.32782/2663-5941/2022.5/14].

[13] Schubert L.: Computational Linguistics. Edward N. Zalta (ed.): The Stanford Encyclopedia of Philosophy, 2020.

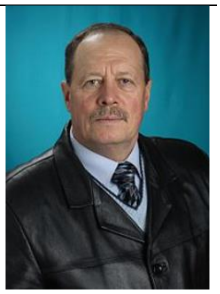
[14] State Standard of Ukraine ITU-R BS.644-1:2005 Audio Broadcasting. Quality indicators of audio signal in the chain of high-quality broadcasting of speech programs. General technical requirements and measurement methods (ITU-R BS.644-1:1990, IDT). State Standard of Ukraine, Kyiv 2007.

[15] Strods D., Smeaton A.: Enhancing Gappy Speech Audio Signals with Generative Adversarial Networks. 34th Irish Signals and Systems Conference – ISSC, 2023, 1–6 [https://doi.org/10.1109/ISSC59246.2023.10161997].

D.Sc., Prof. Olexiy D. Azarov
e-mail: azarova.olexiy@gmail.com

Doctor of Technical Sciences, professor, Head of Department of Computer Engineering of Vinnytsia National Technical University, Vinnytsia, Ukraine. Author of 845 scientific publications, including 22 monographs, 195 scientific articles in professional journals, 33 of which are published in journals of Scopus and Web of Science scientific databases, 571 patents.

https://orcid.org/0000-0002-8501-1379



D.Sc., Prof. Larysa Y. Azarova
e-mail: azarova.larusa@gmail.com

Doctor of Philological Science, professor, Head of Department of Linguistics of Vinnytsia National Technical University, Vinnytsia, Ukraine. Author of 290 scientific publications, including 10 monographs, 212 scientific articles in professional journals, 8 of which are published in journals of Scopus and Web of Science scientific databases.

https://orcid.org/0000-0002-2631-8151

D.Sc., Prof., Corresponding Member NAS of Ukraine Iurii Krak
e-mail: iurii.krak@knu.ua

Doctor of Physics and Mathematical Sciences, professor, Head of Department of Theoretical Cybernetics Taras Shevchenko National University of Kyiv, principal researcher Laboratory of Communicative Information Glushkov Institute of Cybernetics, Kyiv, Ukraine. Author of 726 scientific publications, including 20 monographs, 250 scientific articles in professional journals, 138 of which are published in journals of Scopus and Web of Science scientific databases, 7 patents.

https://orcid.org/ 0000-0002-8043-0785

Ph.D. Leonid V. Krupelnitskyi
e-mail: krup@vntu.edu.ua

Ph.D. in technique, associate professor of Department of Computer Engineering, Vinnytsia National Technical University, Vinnytsia, Ukraine. Developer of analog-digital systems with 30 years of experience. Author of over 100 publications, including 1 textbook, 4 educational manuals, 2 monographs, 55 patents for inventions, more than 22 scientific articles in professional journals, 5 of which are published in journals of Scopus and Web of Science scientific databases.

https://orcid.org/0000-0001-7370-9772

Ph.D. Anzhelika A. Azarova
e-mail: azarova.angelika@gmail.com

Ph.D. in technique, professor of Department of Management and Information Security of Vinnytsia National Technical University, Vinnytsia, Ukraine. Author of more than 300 scientific publications, including 27 monographs, 130 scientific articles in professional journals, 14 of which are published in journals of Scopus and Web of Science scientific databases, 15 textbooks.

https://orcid.org/0000-0003-3340-5701

Veronika V. Azarova
e-mail: nikolazarova14@gmail.com

Student of Borys Grinchenko Kyiv University, Kyiv, Ukraine. Author of 10 scientific publications, including 1 foreign monograph, 2 scientific articles published in journals of Scopus and Web of Science scientific databases.

https://orcid.org/0000-0001-7441-5725