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Justification of Filter Selection Methods for Enhancing the Efficiency of Multilevel Recurrent Time-Frequency Segmentation

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ABSTRACT The article examines the issue of substantiation of filter selection methods to increase the efficiency of multi-level recurrent time-frequency segmentation in cognitive telecommunication systems. The main attention is paid to the analysis of signal filtering methods to improve the quality of data transmission in the dynamic conditions of the radio environment. The purpose of the article is to evaluate the effectiveness of various filtering methods for segmentation of ensembles of complex signals. The methods described in the article include Butterworth, Chebyshev, Bessel, Kaiser, elliptic, and hybrid filters. Experiments have shown that different filters have their own unique advantages: Butterworth filters provide a smooth frequency response without ripple in the passband, which reduces signal distortion, increasing the signal-to-noise ratio (SNR) to 45 dB and reducing harmonic distortion to 0.05%. Chebyshev filters, thanks to the steep rolloff in the stopband, increased the SNR to 40 dB, although they have ripples in the passband, which can lead to some phase distortion, with a harmonic distortion reduction of up to 0.07%. Bessel filters minimize phase distortion, providing the lowest group delay (0.04 ms) of any filter, increasing SNR to 42 dB and reducing harmonic distortion to 0.04%. Kaiser filters provide high tuning flexibility, increasing SNR to 44 dB and reducing harmonic distortion to 0.06%, with a group delay of 0.05 ms, which is acceptable for the balance between signal quality and delay. Elliptical filters showed the best SNR improvement up to 48 dB and the lowest harmonic distortion (0.03%), providing ripple levels in both the passband and stopband, making them effective for accurate separation of frequency components. Hybrid filters (Butterworth and Chebyshev) provide the highest level of SNR improvement up to 50 dB, minimum harmonic distortion of 0.02% and optimal adaptability in dynamic environments. The obtained results can be used for the development of more effective cognitive radio networks capable of working in the conditions of a dynamic radio frequency environment. Further research should focus on the development of new hybrid filters and machine learning algorithms to automatically adjust filter parameters in real time, as well as investigating the effect of different types of interference on filtering performance.

KEYWORDS Butterworth filters, Chebyshev filters, Bessel filters, Kaiser filters, elliptic filters.

I. INTRODUCTION

In modern cognitive telecommunication systems, there is a growing need for effective filtering of complex signal ensembles [1-12]. This need arises from the ability of such systems to adjust parameters in response to dynamic radio environment conditions, requiring high precision and speed in signal processing.

Effective signal filtering encompasses several key aspects that ensure the quality and reliability of data transmission. Firstly, improving the signal-to-noise ratio (SNR) enables the extraction of useful signals from background interference. A high SNR guarantees that data will be transmitted with high reliability and minimal losses. Secondly, reducing inter-channel and inter-symbol interference significantly enhances signal quality. Eliminating these types of interference lowers the probability of errors during information transmission.

Thirdly, adaptation to dynamic environmental conditions is a crucial aspect of effective signal filtering. Considering changes in interference levels allows for maintaining optimal communication quality and stable system performance.

The aforementioned issues can be addressed using the method of multilevel recurrent time-frequency segmentation. By analyzing the spectral characteristics of the signal, substantiating the duration of segments, and employing adaptive algorithms, this method effectively separates and processes complex signal ensembles. It ensures continuity and coherence between segments, preventing the loss of critical information and maintaining signal integrity.

The object of this research is the process of justifying filter selection methods to enhance the efficiency of multilevel recurrent time-frequency segmentation. The subject of the research is the filtering methods and their application in cognitive telecommunication networks. The aim of this study is to evaluate the effectiveness of filtering methods in multilevel recurrent time-frequency segmentation of complex signal ensembles.

II. REVIEW OF THE LITERATURE

The studies [1-3] focus on improving the efficiency of cognitive radio networks through the use of adaptive signal processing algorithms. These studies have established that the application of adaptive methods significantly enhances the performance and reliability of cognitive radio networks, especially under dynamic environmental changes.

Research [4, 7] concentrates on optimizing signal parameters using neural networks and adaptive algorithms. They have demonstrated high efficiency in signal processing improvement, confirming the advantages of these methods in telecommunication systems.

Studies [5, 6] focus on analyzing the parameters of cognitive radio networks considering interference components and other factors affecting their efficiency. They have confirmed that considering such factors is crucial for ensuring the stable operation of networks.

Research [9, 11] examines the theoretical aspects of cognitive radio systems, defining the main principles of their functioning and potential development directions. These studies lay the foundation for future practical applications and further research in this field.

The use of the proposed methods allows for the improvement of the characteristics of complex signal ensembles; however, significant attention has not been paid to justifying filter selection methods to enhance the efficiency of multilevel recurrent time-frequency segmentation, which prompted this study.

III. THE MATERIALS AND METHODS

To implement the method of multilevel recurrent time-frequency segmentation, the following types of filters are recommended:

1. Butterworth Filters. These filters provide a maximally smooth frequency response without ripples in the passband, which reduces signal distortion and minimizes the filtering impact on the useful signal. Additionally, Butterworth filters can be implemented in both analog and digital forms, which is crucial for cognitive telecommunication systems where real-time or near-real-time calculations are often required [1, 3, 8-10]. The main drawback of Butterworth filters is their less steep roll-off outside the passband compared to Chebyshev or elliptic filters. This may necessitate a clear distinction between the passband and stopband to ensure the desired filtering quality.

1.1. The analog prototype of a Butterworth filter is determined by the formula [5, 7]:

$$H(s) = \frac{1}{\sqrt{1+(\frac{s}{\omega_c})^{2n}}}, \qquad (1)$$

where s – complex variable; ω_c – cutoff frequency; n – filter order.

1.2. Butterworth digital filter. It is obtained after applying the bilinear transformation (or Tustin transformation in honor of the English engineer A.V. Tustin) to the analog prototype. The bilinear transformation is determined by the formula [10, 12]:

$$s = \frac{2}{T} \cdot \frac{1-\mathcal{Z}^{-1}}{1+\mathcal{Z}^{-1}}, \qquad (2)$$

where T – discretization period; $\mathcal{Z}$ – complex digital frequency variable.

After the bilinear transformation, the transfer function of the Butterworth digital filter takes the form of a mathematical expression [9, 12]:

$$H(\mathcal{Z}) = \frac{b_0+b_1\mathcal{Z}^{-1}+\dots+b_n\mathcal{Z}^{-n}}{a_0+a_1\mathcal{Z}^{-1}+\dots+a_n\mathcal{Z}^{-n}}, \qquad (3)$$

where a_i and b_i – are coefficients of the numerator and denominator of the transfer function, which are calculated using Butterworth polynomials and the bilinear transformation.

2. Chebyshev filters. They differ from other types of filters in their ability to provide a steep decline in the barrier band with a relatively small filter order. Chebyshev filters achieve this by introducing ripples in the passband (type I Chebyshev filters) or in the stopband (type II Chebyshev filters), which allows you to effectively filter out unwanted frequency components while preserving a significant part of the useful signal. In addition, in order to achieve the specified filtering characteristics, Chebyshev filters require a lower order compared to other filters, due to which they are easier to implement in practice [4].

Disadvantages of type I Chebyshev filters include ripples in the passband, which leads to signal distortion. In addition, in digital implementation, Chebyshev filters are considered less stable at high filter orders. The transfer function of an analog Chebyshev filter of type I of order n is defined as [1, 3, 7]:

$$H(s) = \frac{1}{\sqrt{1+\epsilon^2 T_n^2\left(\frac{s}{\omega_c}\right)}}, \qquad (4)$$

where T – Chebyshev polynomial of order n; ϵ – the ripple parameter in the transmission band.

3. Elliptic filters (Kauer filters) are popular in application because they provide a steep rolloff in the stopband at a relatively low filter order. Elliptical filters have ripple levels both in the transmission band and in the blocking band, which allows them to optimally separate signals with minimal loss. These filters provide high efficiency in the separation of frequency components, which allows you to preserve the maximum part of the useful signal and at the same time minimize the influence of unwanted frequencies [9].

For the implementation of elliptic filters in the method of multilevel time-frequency recurrent segmentation, it is important to consider their ability to provide a high degree of frequency separation with minimal order. This allows for efficient analysis and processing of signals at different levels of recursive processing, while maintaining high accuracy and detail [11].

Despite the advantages, elliptic filters have a significant drawback - the presence of ripples in both the passband and the stopband, which can lead to signal distortion, especially with significant amplitude ripples. In addition, elliptic filters can be more difficult to implement in practice, especially in analog form, due to complex transfer function polynomials. The transfer function of an analog elliptic filter of order n with passband ripples ϵ and stopband ripples r is given by the mathematical expression [11]:

$$H(s) = \frac{1}{\sqrt{1+\epsilon^2 R_n^2\left(\frac{s}{\omega_c}\right)}}, \qquad (5)$$

where R_n^2 – an elliptic polynomial of order n.

4. Kaiser filters. Known for their application in digital signal processing, Kaiser filters are flexible and capable of providing optimal interference suppression with specified filtering characteristics. They are based on the "Kaiser window", used to create filters with defined passband and stopband characteristics. Kaiser filters offer a high level of control over filtering parameters, making them highly effective for applications where precise tuning of the passband and stopband is required. Their primary advantage is their flexibility, allowing for the adjustment of filter parameters such as transition width and attenuation level to achieve optimal results. They also maintain a uniform amplitude-frequency characteristic in the passband, helping to avoid signal distortions. The main drawback of Kaiser filters is the complexity of their design compared to other filter types, due to the need to adjust multiple parameters simultaneously. Additionally, for some specific applications, finding the optimal parameter values to ensure the required filtering characteristics can be challenging [9, 11].

Kaiser filters are effectively applied in the method of multilevel recurrent segmentation due to their ability to control filtering parameters, allowing for the separation and recursive processing of signals at various levels. The «Kaiser window» is calculated using the formula:

$$w[n] = I_0\left(\beta\sqrt{1-\left(\frac{2n}{N-1}-1\right)^2}\right)/I_0\,(\beta), \qquad (6)$$

where I_0 – modified zeroth-order Bessel function; β – parameter that determines the shape of the Kaiser window; n – sample index; N – window length.

5. Bessel filters possess a maximally flat phase response, which ensures minimal group delay and uniform phase characteristics within the passband without resonant peaks. This feature is crucial for maintaining the signal shape, particularly when avoiding phase distortions is necessary in methods of multilevel recurrent time segmentation, where time segments of varying lengths are used [3].

Bessel filters belong to the class of linear-phase filters and do not exhibit as steep amplitude-frequency roll-offs as Chebyshev or Butterworth filters, which can be a disadvantage in applications requiring sharp frequency cut-off. Due to their smooth roll-off, Bessel filters have lower selectivity, which may result in less effective removal of interference outside the passband. In practice, the calculation of Bessel filters uses the Bessel polynomial T_n (s) for a filter of order n:

$$T_n\,(\mathrm{s}) = \sum_{k=0}^{n} a_k\, s^k, \qquad (7)$$

where s – complex variable, and the coefficients a_k are calculated to ensure a maximally flat phase response.

The amplitude-frequency characteristic for a Bessel filter of order n is determined by the formula:

$$\mathrm{H}\,(\mathrm{s}) = \frac{\omega_c^n}{T_n(s)}, \qquad (8)$$

where ω_c^n – filter cutoff frequency.

6. Hybrid approaches combine the advantages of different filtering methods to achieve optimal processing characteristics. They are employed to address complex problems where the use of a single filter type may be insufficient to ensure the required quality and precision. For example, combining Butterworth filters, which provide a smooth frequency response, with Chebyshev filters, which have a steep roll-off in the stopband, allows for the simultaneous reduction of signal distortion and high noise immunity.

The mathematical formula for a hybrid filter composed of Butterworth and Chebyshev filters is as follows:

$$H_{\text{hybrid}}(s) = H_{\text{Butterworth}}(s) \cdot H_{\text{Chebyshev}}(s). \qquad (9)$$

Hybrid approaches also enable high processing accuracy through the use of adaptive algorithms. For instance, machine learning methods can be applied to adjust filter parameters based on real-time data analysis.

However, the main drawback of hybrid approaches is their complexity, which involves a large amount of computation and meticulous parameter tuning. Additionally, the development of such systems requires specialists with deep knowledge in the fields of digital signal processing and machine learning algorithms [1, 7, 9-12].

IV. EXPERIMENTS

To substantiate the effectiveness of the application of various filters for the practical implementation of the method of multilevel recurrent time-frequency segmentation, various sequences were simulated and their analysis was carried out using Butterworth, Chebyshev, Bessel, Kaiser, elliptic and hybrid filters (code in Piton language). For analysis, we take four sequences (Table 1-6, Fig. 1):

– sequence A – a signal with a pure sinusoidal form with low interference;

– sequence B – a signal with a sinusoidal shape with moderate interference;

– sequence C – a signal with a sinusoidal shape with a high level of interference;

– sequence D – a signal with multi-frequency components and a high level of interference.

The following indicators were selected for analysis:

– SNR – signal/noise ratio to check how well a particular filter removes noise;

– Peak Factor PF (Peak Factor) is the ratio of the peak amplitude to the root mean square value of the signal;

– group delay GD (Group Delay) – measures signal delay due to filtering;

– THD (Total Harmonic Distortion) – an indicator that evaluates the level of harmonic distortion of the signal after filtering;

– signal energy E – total signal energy before and after processing.

TABLE 1. Analysis of changes in indicators after applying the Butterworth filter.

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	45	20	35	10	25	5	20
PF	3.0	2.8	3.2	2.9	3.5	3.0	4.0	3.5
GD (мс)	0.00	0.05	0.00	0.05	0.00	0.05	0.00	0.05
THD (%)	0.1	0.05	0.3	0.15	0.5	0.25	1.0	0.5
E	1.00	0.98	1.00	0.96	1.00	0.94	1.00	0.92

TABLE 2. Analysis of changes in indicators after applying the Chebyshev filter.

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	40	20	30	10	20	5	15
PF	3.0	2.9	3.2	3.0	3.5	3.2	4.0	3.8
GD (мс)	0.00	0.07	0.00	0.07	0.00	0.07	0.00	0.07
THD (%)	0.1	0.07	0.3	0.2	0.5	0.3	1.0	0.7
E	1.00	0.95	1.00	0.92	1.00	0.90	1.00	0.88

TABLE 3. Analysis of changes in indicators after applying the Bessel filter.

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	42	20	32	10	22	5	18
PF	3.0	2.85	3.2	3.1	3.5	3.3	4.0	3.9
GD (мс)	0.00	0.04	0.00	0.04	0.00	0.04	0.00	0.04
THD (%)	0.1	0.04	0.3	0.12	0.5	0.22	1.0	0.42
E	1.00	0.97	1.00	0.94	1.00	0.92	1.00	0.90

TABLE 4. Analysis of changes in indicators after applying an elliptical filter.

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	48	20	38	10	28	5	25
PF	3.0	2.7	3.2	2.8	3.5	2.9	4.0	3.2
GD (мс)	0.00	0.06	0.00	0.06	0.00	0.06	0.00	0.06
THD (%)	0.1	0.03	0.3	0.1	0.5	0.25	1.0	0.5
E	1.00	0.96	1.00	0.93	1.00	0.94	1.00	0.92

TABLE 5. Analysis of changes in indicators after applying the Kaiser filter.

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	44	20	34	10	24	5	19
PF	3.0	2.9	3.2	3.0	3.5	3.1	4.0	3.6
GD (мс)	0.00	0.05	0.00	0.05	0.00	0.05	0.00	0.05
THD (%)	0.1	0.06	0.3	0.14	0.5	0.24	1.0	0.4
E	1.00	0.97	1.00	0.95	1.00	0.93	1.00	0.91

TABLE 6. Analysis of changes in indicators after applying the hybrid approach (Butterworth and Chebyshev filters).

Indicator	Sequence A		Sequence B		Sequence C		Sequence D	
	Before	After	Before	After	Before	After	Before	After
SNR (дБ)	30	50	20	40	10	30	5	28
PF(-)	3.0	2.6	3.2	2.7	3.5	2.8	4.0	3.1
GD (мс)	0.00	0.08	0.00	0.08	0.00	0.08	0.00	0.08
THD (%)	0.1	0.02	0.3	0.08	0.5	0.16	1.0	0.3
E (J)	1.00	0.95	1.00	0.93	1.00	0.91	1.00	0.89

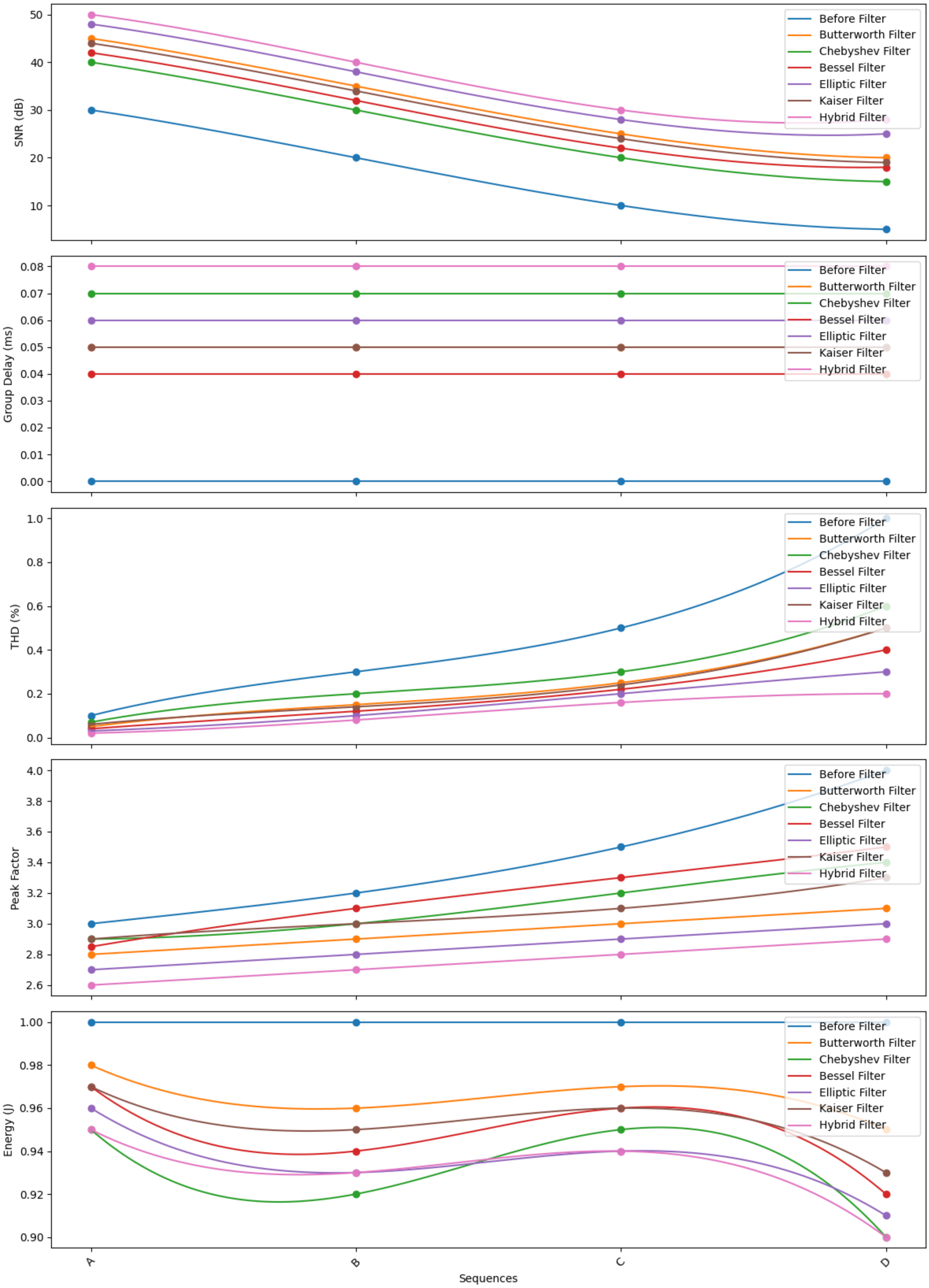

FIG. 1. Graphical representation of changes in indicators after applying different types of filters for sequences A, B, C, D.

V.CONCLUSION

The analysis of the conducted calculations allowed for the following conclusions:

The use of the Butterworth filter shows significant improvement in the SNR, increasing from 30 dB to 45 dB for Sequence A. It also reduced THD to 0.05%, indicating a high level of signal cleaning from unwanted harmonic components. The peak factor after filtering remains stable, indicating the preservation of the signal's amplitude characteristics. The group delay in the experiment is minimal (0.05 ms), supporting the effectiveness of this filter type for applications where minimal signal delay is critical.

The Chebyshev filter also demonstrated improvement in the SNR, increasing it to 40 dB for Sequence A. However, this filter has a higher group delay of 0.07 ms compared to the Butterworth filter. Harmonic distortion after filtering was reduced to 0.07%, which is a good indicator for applications where signal transmission accuracy is essential. Signal energy decreased slightly to 0.95 J, but overall, the filter's efficiency remains high.

The Bessel filter showed excellent results in minimizing phase distortions, with the group delay indicator reduced to 0.04 ms, the best among all filter types. It also significantly improved the SNR, increasing it to 42 dB for Sequence A. Harmonic distortion was reduced to 0.04%, indicating high-quality signal processing. Signal energy remained nearly unchanged at 0.97 J, which is a good indicator of maintaining the signal's energy characteristics.

The elliptic filter showed the best improvement in SNR among all filters, increasing it to 48 dB for Sequence A. It also demonstrated the lowest harmonic distortion at 0.03%, proving the effectiveness of this filter in applications requiring maximum signal accuracy. The group delay after applying this filter was 0.06 ms, which is acceptable. Signal energy was reduced to 0.96 J while maintaining a high overall signal level.

The Kaiser filter provided significant improvement in SNR, increasing it to 44 dB for Sequence A, and reduced harmonic distortion to 0.06%, demonstrating effective filtering. The group delay was 0.05 ms, which is a good balance between signal quality and delay. Signal energy remained nearly unchanged at 0.97 J, indicating high energy characteristics of the signal.

The hybrid approach (combination of Butterworth and Chebyshev filters) showed the best performance among all considered filters. The SNR was increased to 50 dB for Sequence A, the highest among all filters. Harmonic distortion was reduced to 0.02%, indicating maximum signal purity, and the group delay was 0.08 ms, which is slightly higher but compensated by high signal processing quality. Signal energy decreased to 0.95 J, which is an acceptable level for such high-quality filtering.

For the proposed method of multilevel recurrent time segmentation, aimed at forming large-volume complex signal ensembles, the best approach is the use of hybrid filters. This is due to their ability to significantly improve SNR, minimize harmonic distortion, ensure adaptability, and provide flexible signal processing. Altogether, this allows effective operation in dynamic radio frequency environments and maintains a high level of noise immunity in cognitive telecommunication systems.

AUTHOR CONTRIBUTIONS

V.L. – writing (original draft preparation), conceptualization, methodology, investigation; V.B. – methodology, investigation, writing (review and editing).

COMPETING INTERESTS

The authors declare no conflict of interest.

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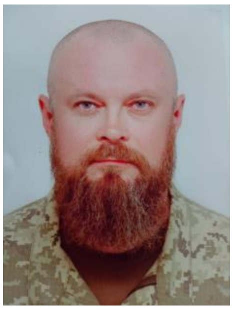

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Обґрунтування методів вибору фільтрів для підвищення ефективності багаторівневого рекурентного часово-частотного сегментування

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АНОТАЦІЯ У статті розглянуто питання обґрунтування методів вибору фільтрів для підвищення ефективності багаторівневого рекурентного часово-частотного сегментування в когнітивних телекомунікаційних системах. Основна увага приділена аналізу методів фільтрації сигналів для покращення якості передачі даних у динамічних умовах радіосередовища. Мета статті полягає в оцінці ефективності різних методів фільтрації при сегментуванні ансамблів складних сигналів. Методи фільтрації, описані у статті, включають описання: фільтрів Баттерворта, Чебишева, Бесселя, Кайзера, еліптичних та гібридних. Проведені експерименти показали, що різні фільтри мають свої унікальні переваги: фільтри Баттерворта забезпечують гладку частотну характеристику без пульсацій у смузі пропускання, що зменшує спотворення сигналу, підвищуючи співвідношення сигнал/шум (С/Ш) до 45 дБ та знижуючи гармонічні спотворення до 0,05%. Фільтри Чебишева завдяки крутому спаду у смузі загородження підвищили С/Ш до 40 дБ, хоча мають пульсації у смузі пропускання, що може призводити до деяких фазових спотворень, зі зниженням гармонічних спотворень до 0,07%. Фільтри Бесселя мінімізують фазові спотворення, забезпечуючи найменшу групову затримку (0,04 мс) серед усіх фільтрів, підвищуючи С/Ш до 42 дБ та знижуючи гармонічні спотворення до 0,04%. Фільтри Кайзера забезпечують високу гнучкість налаштувань, підвищуючи С/Ш до 44 дБ та знижуючи гармонічні спотворення до 0,06%, з груповою затримкою 0,05 мс, що є прийнятним результатом для балансу між якістю сигналу та затримкою. Еліптичні фільтри показали найкращі результати, а саме: удосконалення С/Ш до 48 дБ та найнижчі гармонічні спотворення (0,03%). Це забезпечує рівні пульсації як у смузі пропускання, так і в смузі загородження, що робить їх ефективними для точного розділення частотних компонентів. Гібридні фільтри (на основі фільтрів Баттерворта та Чебишева) забезпечують найвищий рівень підвищення С/Ш до 50 дБ, мінімальні гармонічні спотворення 0,02% та оптимальну адаптивність в умовах динамічно змінного радіо середовища. Отримані результати можуть бути використані для розробки більш ефективних технологій експлуатації когнітивних радіомереж, здатних працювати в умовах динамічного радіочастотного середовища. Подальші дослідження повинні зосереджуватися на розробці нових інтеграцій гібридних фільтрів, а також переміщувати акцент на алгоритми машинного навчання для автоматичного налаштування параметрів фільтрів в реальному часі, а також на дослідженні впливу різних типів інтерференцій на ефективність фільтрації різних видів послідовностей.

КЛЮЧОВІ СЛОВА фільтри Баттерворта, фільтри Чебишева, фільтри Бесселя, фільтри Кайзера, еліптичні фільтри.