UKLANJANJE NEŽELJENOG SIGNALA IZ DIGITALNOG AUDIO ZAPISA
REMOVAL OF OBSTRUCTIVE SIGNAL FROM DIGITAL AUDIO RECORDING

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Sažetak: *Snimljeni audio signal pored željenog signala često sadrži i neželjeni signal koji mu je pridodan. U ovom je radu prikazan postupak projektiranja digitalnih filtara s konačnim impulsnih odzivom čija je svrha potiskivanje neželjenog signala.*

Ključne riječi: - digitalni audio signal
                   - filtri s konačnim impulsnim odzivom
                   - Parks-McClellanov algoritam

Summary: *Very often, a recorded audio signal contains an added obstructive signal. This paper presents a method for designing digital filters of finite impulse response which are used in order to suppress an obstructive signal.*

Key words: - digital audio signal
               - finite impulse response filters
               - Parks-McClellan algorithm

1. UVOD

Prilikom snimanja određenog audio zapisa čest je slučaj da je originalnom signalu pridodan i neželjeni signal. Ako nije predviđena mogućnost ponovnog snimanja željenog audio signala, potrebno je izdvojiti i eliminirati neželjeni signal. Pod uvjetom da su poznata svojstva neželjenog signala, njegovo se uklanjanje može izvršiti digitalnom filtracijom s odgovarajućim filtrima. U ovom je radu prezentiran postupak projektiranja filtara temeljen na Parks-McClellanovom algoritmu [1]. Pretpostavljeno je da je spektar neželjenog signala poznat i lokaliziran u frekvenciji.

U ovom je radu predložen postupak određivanja filtarske specifikacije temeljen na proračunu po dijelovima konstante aproksimacije amplitudnog spektra neželjenog signala (u daljnjem tekstu, kontura spektra). Prototipni je filtar dobiven korištenjem konture spektra i praga čujnosti kao granične vrijednosti konture spektra signala. Područje gušenja prototipnog filtra predstavlja područje frekvencija za koje je kontura spektra neželjenog signala veća od praga čujnosti. Filtar za uklanjanje neželjenog signala dobiven je na temelju prototipnog filtra uz korištenje Parks-McClellanovog algoritma. Vrijednost praga čujnosti može se koristiti za podešavanje širine područja gušenja projektiranog filtra.

1. INTRODUCTION

During the audio recording process an obstructive signal can be accidentally added to the original signal. If a repeatition of the recording process is not possible, it is necessary to isolate and eliminate the obstructive signal. If the properties of a signal are known, its removal can be performed by means of digital filtering with the appropriate filters. In this paper we present a filter design procedure that is based on the Parks-McClellan algorithm [1]. It is assumed that the spectrum of the obstructive signal is known and localized in the frequency.

In this paper, we propose a procedure for determining a filter specification, on a piece-wise basis with constant approximation of the obstructive signal's amplitude spectrum (called *spectral outline* in the following text). A prototype filter is obtained by using the spectral outline and an audibility threshold, which represents a critical value of the spectral outline. The stopband of a prototype filter represents a frequency band for which a spectral outline of an obstructive signal is greater than the audibility threshold. A filter for removing the obstructive signal is obtained based on the prototype filter, by using the Parks-McClellan algorithm. The audibility threshold can be used for tuning the stopband of the filter.

2. MOTIVACIJA – SPEKTRALNA SVOJSTVA NEŽELJENOG SIGNALA

Digitalni audio signal *y* može se prikazati kao zbroj željenog audio signala *x* i neželjenog audio signala *v*:

2. MOTIVATION – SPECTRAL PROPERTIES OF THE OBSTRUCTIVE SIGNAL

The digital audio signal *y* can be represented as a sum of an original signal *x* and an obstructive signal *v*:

$$y[n] = x[n] + v[n] \tag{1}$$

U ovom je radu pretpostavljeno poznavanje spektralnih svojstava neželjenog signala, kojeg je potrebno ukloniti iz ukupnog signala. Na Slici 1 prikazani su primjeri valnih oblika željenog i neželjenog zvučnog signala te ukupnog signala u trajanju od 1 sekunde. Željeni zvučni zapis pripada isječku glazbe iz filma, a neželjeni audio signal predstavlja zvuk zvona telefona.

In this paper, it is implied that the spectral properties of the obstructive signal that should be removed from the total signal are known. Figure 1 gives examples for waveforms of the original, obstructive and total signals lasting one second. The original signal is a movie music clip and the obstructive signal is represented as the background ringing of a telephone.

(a)

(b)

(c)
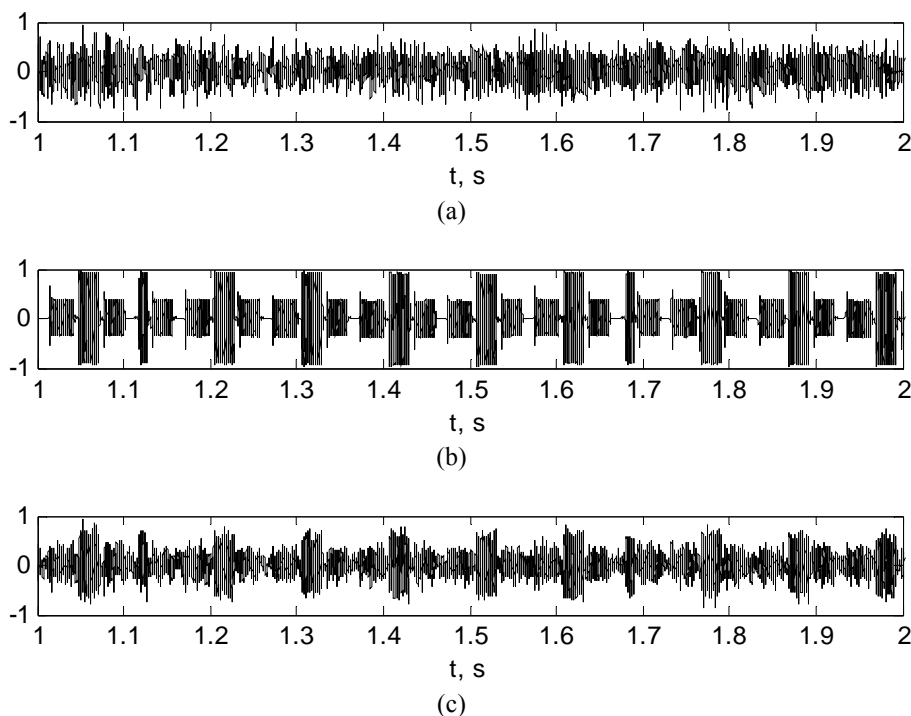
Slika 1.   Valni oblik: a) željenog signala, b) neželjenog signala, c) ukupnog signala
Figure 1. Waveforms of: a) the original signal, b) the obstructive signal, c) the total signal

Slikom 2 prikazani su amplitudni spektri željenog i neželjenog zvučnog signala te ukupnog signala, dobiveni računanjem diskretne Fourierove transformacije (DFT) signala.

Figure 2 shows the magnitude spectra of the original, the obstructive and the total signal obtained by calculating a discrete Fourier transform (DFT).

$$X[k] = \sum_{n=0}^{N-1} x[n] \cdot e^{-j\frac{2\pi}{N}nk} \tag{2}$$

Slika 2 prikazuje normirane vrijednosti DFT-a dobivene kao

Figure 2 shows the normalized values of DFT obtained as

$$A = 20\log_{10}\left(\frac{|X[k]|}{N/2}\right) \tag{3}$$

Ovo normiranje rezultira time da 0 dB odgovara sinusnom signalu amplitude 1. Na *x*-osi označene su odgovarajuće frekvencije u rasponu do polovice frekvencije uzorkovanja signala, koja iznosi 44100 Hz za dani primjer.

Kako se može vidjeti sa slika, spektar neželjenog signala koncentriran je na frekvencijama između 1.5 kHz i 3 kHz pa zaključujemo da filtar za uklanjanje neželjenog signala mora imati područje gušenja upravo u navedenom intervalu.

The normalization gives 0 dB for a sinusoidal signal of 1. On the x-axis, corresponding frequencies are given in the range of up to half of the sampling frequency, which is 44100 Hz for the given example.

As it can be seen, the magnitude spectrum of the obstructive signal is concentrated in the 1.5 to 3 kHz frequency band so we can conclude that the filter that will be used for removing the obstructive signal should have the stopband in the same frequency band.
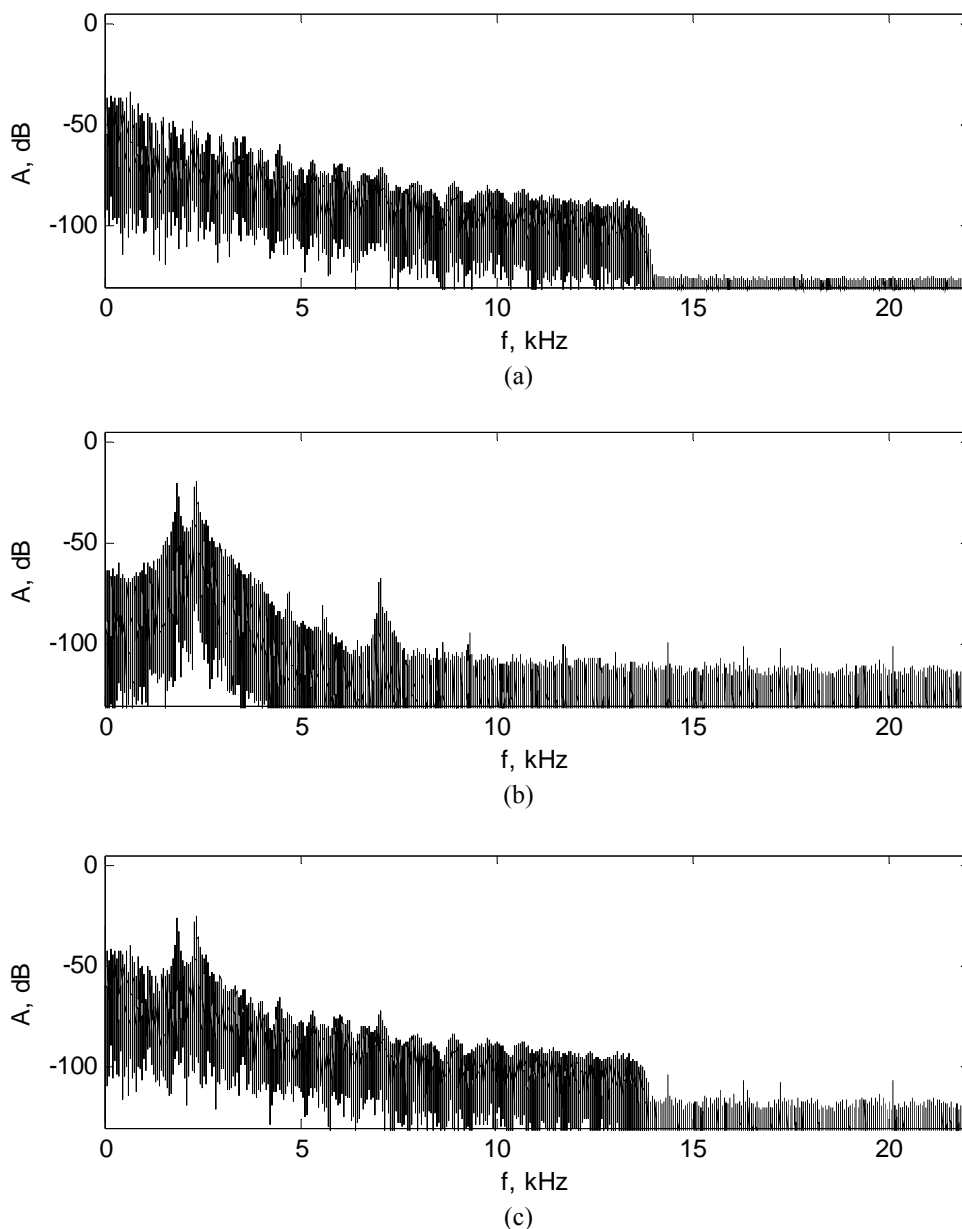
Slika 2.   Amplitudni spektar: a) željenog signala, b) neželjenog signala, c) ukupnog signala
Figure 2. Magnitude spectra of: a) the original signal, b) the obstructive signal, c) the total signal

U nastavku je predložen algoritam koji na osnovu amplitudnog spektra neželjenog audio signala određuje karakteristiku prototipnog filtra koji prigušuje taj signal. Projektirani filtar ne mora imati samo jedno područje gušenja kao u navedenom primjeru, već više njih, ovisno o spektru neželjenog signala.

In the next section, we propose an algorithm which uses the spectrum of the obstructive signal in order to determine a prototype filter which attenuates it. The designed filter need necessarily have only a single stopband, it can have more than one stopband, depending on the spectrum of the signal.

3. PROJEKTIRANJE FILTARA

3. FILTER DESIGN

Na temelju vrijednosti uzoraka DFT-a neželjenog signala $V[k]$ određuje se frekvencijska karakteristika prototipnog filtra. Svrha projektiranog filtra je čim bolje ukloniti neželjeni signal i u isto vrijeme što je moguće manje degradirati željeni signal. U tu svrhu, filtar mora imati što je moguće uža područja gušenja, koja odgovaraju područjima dominantne energije spektra neželjenog signala. Kako je prikazano na Slici 2, spektar signala postoji u svim područjima frekvencija, ali s različitom amplitudom.

Kako područja gušenja filtra ne bi bila nepotrebno široka, određuje se prag čujnosti, odnosno zadovoljavajući faktor prigušenja od 45 dB. Naime, signal stišan za 45 dB gotovo se više ni ne čuje.

Based on the DFT samples $V[k]$ of the obstructive signal, a frequency response of the prototype filter is determined. The purpose of the filter design is to remove the obstructive signal as much as possible and at the same time not to degrade the original signal. Therefore, the filter has to have as narrow stopbands as possible, which correspond to the frequency bands in which the obstructive signal spectrum is dominant. In the example seen in Figure 2, the signal spectrum is nonzero for all frequencies, with varying amplitude.

To avoid unnecessarily wide stopbands of the filter, an audibility threshold is determined. An attenuation factor of 45 dB is determined to be satisfactory since a signal attenuated to 45 dB is almost impossible to hear.
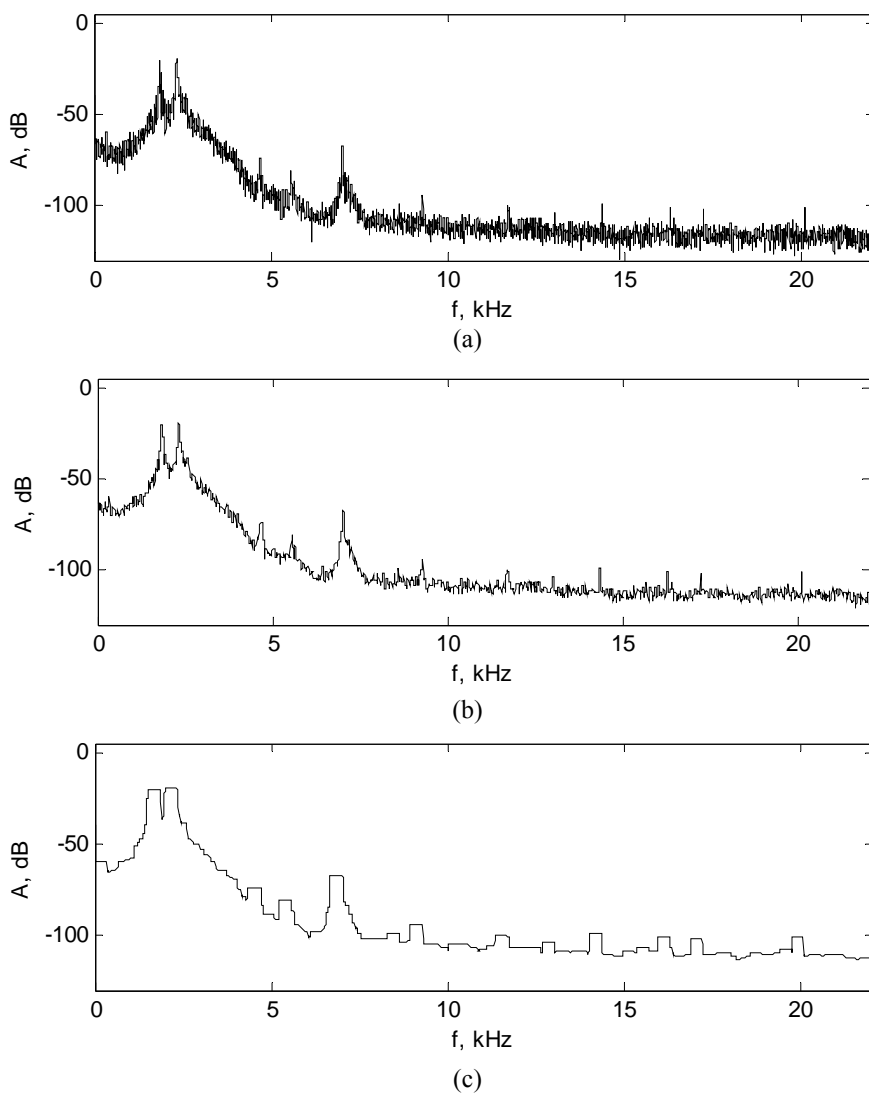
3.1. Kontura spektra neželjenog signala

3.1. Spectral outline of the obstructive signal

Kako je prikazano na Slici 2, vrijednosti DFT-a jako variraju, što bi rezultirao s nepotrebno puno alternirajućih područja gušenja i propuštanja. Da bi se to spriječilo, u ovom je radu predloženo određivanje konture spektra signala na osnovu koje se određuje karakteristika filtra. Kontura spektra je po dijelovima konstantna aproksimacija spektra signala. Dobiva se tako da se za svaki uzorak u frekvenciji uzme maksimalna vrijednosti amplitude spektra njemu susjednih D uzoraka:

As shown in Figure 2 DFT values vary significantly which would lead to an unnecessarily frequent alternation of the passbands and stopbands of the filter. In order to avoid this, in this paper we propose a method for determining the spectral outline of the signal which is used for the filter design. Spectral outline is a piece-wise constant approximation of the signal spectrum. It is obtained for each frequency sample by taking the maximal value D of its neighbouring samples:

$$V_D[k] = \max_{m \in [k, k+D]} \left(|V[m]| \right) \tag{4}$$

Slikom 3 prikazane su konture spektra neželjenog zvuka zvona telefona za različite vrijednosti parametra D. Slika 4 prikazuje prototipni filtar dobiven na temelju presjecišta konture spektra neželjenog signala uz D=1000 i praga čujnosti od -45 dB.
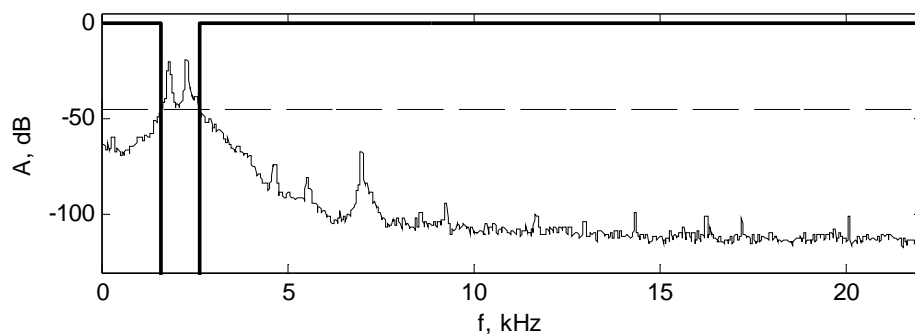
Figure 3 shows the spectral outline of the obstructive signal for different values of D. Figure 4 shows a prototype filter obtained on the basis of the intersection of the spectral outline of the obstructive signal for $D = 1000$ and an audibility threshold of -45 dB.

3.2. Parks-McClellanov algoritam

3.2. Parks-McClellan algorithm

Nakon određivanja konture spektra iz koje slijedi karakteristika prototipnog filtra, pristupa se određivanju koeficijenata impulsnog odziva realnog filtra. Realizirat će se FIR filtri upotrebom Parks-McClellanovog algoritma [1]. U razvoju izraza za željeni filtar analiziran je FIR filtar parnog reda L (neparni broj uzoraka impulsnog odziva) i simetričnog impulsnog odziva s obzirom na nulu (nekauzalni filtar). Frekvencijska je karakteristika takvog filtra realna i ima oblik koji je dan izrazom:

After the spectral outline has been determined, from which the prototype filter's frequency response follows, the next task is to calculate the coefficients of the real filter's frequency response. FIR filters will be realized by using the Parks-McClellan algorithm [1]. In order to derive expressions for desired filters we will use a FIR filter of even order L (an odd number of samples of the corresponding impulse response). The frequency response of such a filter is real and is given by:

Slika 3. Kontura spektra neželjenog signala za: a) D=100, b) D=500, c) D=5000. DFT je računat u N=618086 točaka
Figure 3. Spectral outline of the obstructive signal for: a) D=100, b) D=500, c) D=5000. DFT is calculated in N=618086 points

Slika 4.   Kontura spektra neželjenog signala za D=1000 i rezultirajuća karakteristika prototipnog filtra (deblja linija) dobivena uz prag čujnosti od -45 dB (crtkana linija)
Figure 4. Spectral outline of the obstructive signal for D=1000 and the frequency response of the prototype filter (bold line) obtained with audibility threshold of -45 dB (dashed line)

$$A_e(\omega) = h_e[0] + \sum_{n=1}^{M} 2h_e[n]\cos(\omega n) \tag{5}$$

gdje je $M = L/2$ cijeli broj. Izraz možemo pisati i drukčije kao polinom M-tog reda s obzirom na $\cos(\omega)$:

where $M=L/2$ is an integer. The expression can be written in a different way as the M-th order polynomial with respect to $\cos(\omega)$:

$$A_e(\omega) = \sum_{n=0}^{M} a_n (\cos\omega)^n \tag{6}$$

Parks i McClellan su dokazali da uz fiksirani red filtra i granice područja propuštanja i gušenja, dizajn filtra može biti promatran kroz prizmu Chebyshevljeve aproksimacije na disjunktnim skupovima [2, 3]. Ovaj je problem već duže bio poznat i obrađen u području teorije aproksimacije [4].

U ovom je slučaju problem aproksimacije sveden na određivanje amplitudno-frekvencijske karakteristike $A_e(\omega)$ koja aproksimira željenu frekvencijsku karakteristiku $D(\omega)$. Pogreška aproksimacije definira se kao:

Parks and McClellan have proven that when having a fixed filter order and fixed passband and stopband regions, filter design can be viewed as a Chebyshev approximation problem over disjoint sets [2, 3]. This problem has already been treated and solved in the field of approximation theory [4]. In our case, the approximation problem is reduced to determining the frequency response $A_e(\omega)$ which approximates a desired frequency response $D(\omega)$. The approximation error is defined as:

$$E(\omega) = W(\omega)[D(\omega) - A_e(\omega)] \tag{7}$$

gdje je $W(\omega)$ težinska funkcija. Cilj je odrediti A_e, odnosno pripadajući impulsni odziv h_e koji minimizira najveću pogrešku aproksimacije za definirana područja propuštanja i gušenja projektiranog filtra. Za rješenje ovog problema vrlo je važan teorem alternacije koji kaže da je nužan i dovoljan uvjet da polinom $A_e(\cos(\omega))$ minimizira maksimalnu apsolutnu vrijednost pogreške E taj da funkcija $E(\cos(\omega))$ ima barem $M+2$ alternacije koje naizmjenično poprimaju vrijednosti $+\delta$ i $-\delta$. Iz teorema alternacije slijede jednadžbe:

where $W(\omega)$ is a weight function. The goal is to determine A_e, i.e. the corresponding impulse response h_e which minimizes the maximum weighted approximation error for given passband and stopband regions. In order to solve the problem, the alternation theorem is of critical importance, which states that a necessary and sufficient condition for a polynomial $A_e(\cos(\omega))$ to minimize the maximum error E is that the function $E(\cos(\omega))$ exhibits at least $M+2$ alternations which take alternating values $+\delta$ and $-\delta$. The alternation theorem leads to the following equation:
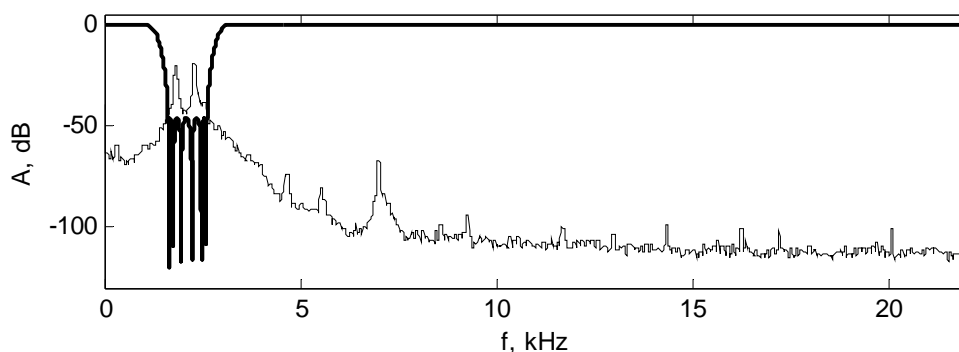
$$W(\omega_i)[D(\omega_i) - A_e(\omega_i)] = (-1)^{i+1} \cdot \delta \quad , \quad i = 1,2,...,(M+2) \tag{8}$$

Za definirani skup kružnih frekvencija ω_i i pripadne $x_i = \cos(\omega_i)$ za koje funkcija pogreške poprima maksimalne apsolutne vrijednosti može se dobiti skup linearnih jednadžbi s nepoznanicama $a_0, a_1, ... a_M$ i δ:

For a given set of frequencies ω_i and corresponding $x_i = \cos(\omega_i)$ for which the error function takes maximal absolute values we can obtain a set of linear equations with unknowns $a_0, a_1, ... a_M$, and δ:

$$\begin{bmatrix} 1 & x_1 & x_1^2 & ... & x_1^M & \dfrac{1}{W(\omega_1)} \\ 1 & x_2 & x_2^2 & ... & x_2^M & \dfrac{-1}{W(\omega_2)} \\ ... & ... & ... & ... & ... & ... \\ 1 & x_{M+2} & x_{M+2}^2 & ... & x_{M+2}^M & \dfrac{(-1)^{M+2}}{W(\omega_{M+2})} \end{bmatrix} \cdot \begin{bmatrix} a_0 \\ a_1 \\ ... \\ \delta \end{bmatrix} = \begin{bmatrix} D(\omega_1) \\ D(\omega_2) \\ ... \\ D(\omega_{M+2}) \end{bmatrix} \tag{9}$$

Dani sustav jednadžbi služi kao temelj iterativnog algoritma koji započinje sa skupom pretpostavljenih točaka alternacije ω_i, a završava s pravim točkama

This set of equations is the basis for an iterative algorithm which starts by guessing a set of alternation frequencies ω_i, and ends with true alternation frequencies

alternacije i koeficijentima filtra [5]. Ovakvo projektiranje filtara omogućava nezavisnu kontrolu nad valovitostima u pojedinim pojasevima i daje filtre nižeg reda za ostvarenje iste željene karakteristike. Implementacija ove metode u MATLAB-u dana je funkcijom *remez*.

and coefficients of the filter [5]. This filter design method allows independent control over ripples in each band and gives lower order filters for the same filter specification. The method is implemented in MATLAB by the *remez* function.

4. REZULTATI

4. RESULTS

Na temelju prototipne filtarske karakteristike sa Slike 4 projektiran je digitalni filtar reda 200. Rezultirajuća amplitudno-frekvencijska karakteristika filtra prikazana je na Slici 5.
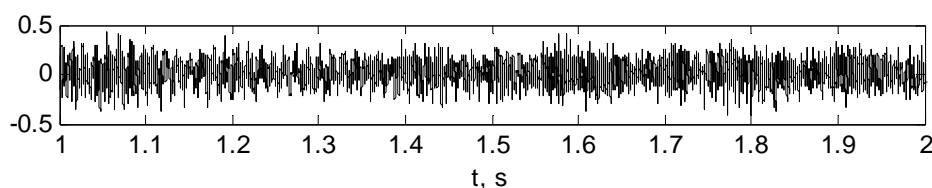
Based on the prototype filter from Figure 4, we have designed a digital filter of the order 200. The obtained magnitude response is shown in Figure 5.

Slika 5.   Karakteristika filtra (deblja linija) i kontura spektra neželjenog signala za D=1000
Figure 5. Magnitude response of the filter (bold line) and the spectral outline of the obstructive signal for D=1000

Da su područja gušenja ili propuštanja bila uža, moglo bi se dogoditi da dolazi do većih odstupanja dobivene filtarske karakteristike od prototipne. To se može ispraviti faktorima težinske funkcije za pojedino područje te promjenom reda filtra, a prototipnoj se karakteristici područje gušenja može popraviti unosom praga čujnosti ili uzoraka glađenja spektra *D*. Time bi i dobivena karakteristika lakše pratila željenu.
Djelovanje dobivenog filtra na ukupni signal prikazano je na slikama 6 i 7.

For the case of narrower passbands or stopbands, greater deviations from the prototype filter specification can be expected. In order to solve the problem, a different choice of weights can be made or the filter order can be increased. Also, the prototype filter can be changed by entering a different audibility threshold or a different parameter *D* which influences the shape of the spectral outline. The results of filtering of the total signal with a designed filter is shown in figures 6 and 7.

Slika 6.   Valni oblik dobiven filtracijom ukupnog signala
Figure 6. Waveform obtained by filtering the total signal

Kako se može vidjeti iz Slike 7, filtriranjem je iz spektra ukupnog signala uklonjeno područje frekvencija kojima dominira spektar neželjenog zvuka zvona telefona.

As shown in Figure 7, filtration removed a frequency band from the total signal which is dominated by the obstructive signal's spectrum, i.e. the spectrum of the phone ring.

Samom je filtracijom oštećen i željeni signal pa dobiveni filtrirani zapis ipak nije istovjetan digitalnom audio zapisu željenog signala. Međutim, preslušavanjem filtriranog audio signala zaključujemo da se originalni signal može dobro čuti, dok je gotovo nemoguće čuti neželjeni signal [6].

The filtration process degraded the original signal too, so the obtained filtered signal is not the same as the original signal. However, after listening to the filtered audio signal, we conclude that the original signal can be heard well, while it is almost impossible to hear the obstructive signal [6].
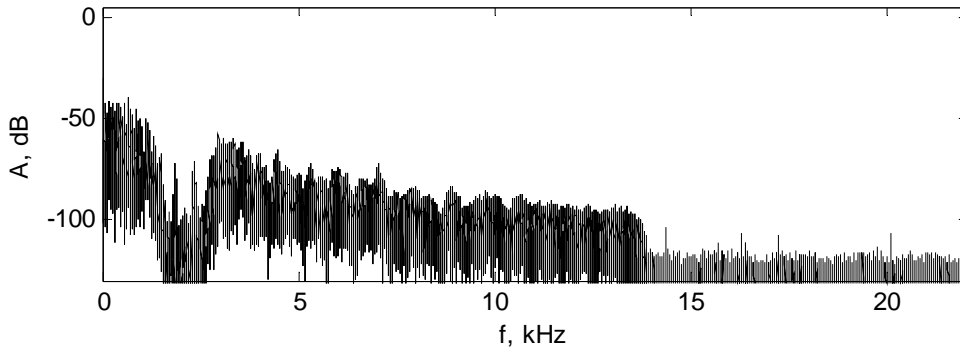
Slika 7.   Spektar filtriranog ukupnog signala
Figure 7. Magnitude spectrum of the filtered total signal

5. ZAKLJUČAK

U radu je predložena metoda uklanjanja neželjenog audio signala iz digitalnog audio zapisa. Predloženi algoritam za neželjeni audio signal projektira filtar s ciljem njegovog uklanjanja iz ukupnog signala.

Kroz primjer uklanjanja zvuka zvona telefona pridodanom isječku filmske glazbe pokazana je zadovoljavajuća kvaliteta potiskivanja neželjenog audio signala uz uhu neprimjetnu degradaciju željenog audio zapisa.

Aplikaciju je moguće unaprijediti kroz automatski izračun parametra D potrebnog za određivanje konture spektra. Parametar D mogao bi biti određen adaptivno, ovisno o varijabilnosti dobivenih uzoraka DFT-a.

Iako je predložena metoda uklanjanja neželjenog audio signala relativno jednostavna, njena je upotrebljivost ograničena. Znatno bolje rezultate moguće je postići korištenjem adaptivnih metoda kao što je upotreba Wienerovog filtra [7]. Ovakve metode uzimaju u obzir ne samo svojstva neželjenog, nego i željenog signala.

5. CONCLUSION

In this paper, we have proposed a method for removing an obstructive audio signal from an audio recording. The proposed algorithm designs a filter for a given obstructive signal with the goal of removing it from the total signal.

We have presented an example of removing a telephone ring from a music clip which resulted in the satisfactory quality of obstructive signal suppression with a degradation of the original audio in to level that is almost undetectable by a typical listener.

Possible improvement of the application can be made in automatic calculation of the parameter D which is needed for creating a spectral outline. The parameter can be adaptively chosen, based on the variability of the DFT samples.

Although the proposed method is relatively simple and straightforward to implement, its applicability is limited. Much better results can be obtained by using adaptive methods such as Wiener filtering [7] that take into account not only the properties of the obstructive signal but the properties of the original signal as well.

6. POPIS OZNAKA

6. LIST OF SYMBOLS

korak u vremenu	n	time index
korak u frekvenciji	k	frequency index
vremenski diskretni signal	$x[n]$	discrete-time signal
DFT vremenski diskretnog signala	$X[k]$	DFT of discrete-time signal
broj uzoraka DFT-a	N	number of samples of DFT
broj uzoraka za izračun frekvencijske ovojnice	D	number of samples for calculation of a spectral outline
impulsni odziv filtra	$h[n]$	impulse response of a filter
maksimalna pogreška aproksimacije filtra	δ	maximal error of filter approximation

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