Mathematical and Computer Modelling 57 (2013) 2660–2670

Contents lists available at SciVerse ScienceDirect

Mathematical and Computer Modelling

journal homepage: www.elsevier.com/locate/mcm

Drift Correction Modulation scheme for digital signal processing

Zbigniew Piotrowski *

Military University of Technology, Kaliskiego 1, PL-01-476 Warsaw, Poland

ARTICLE INFO

Article history:
Received 25 January 2011
Received in revised form 8 September 2011
Accepted 12 September 2011

Keywords:
Drift Correction Modulation
DCM
Drift scanner method
Normalized data payload
Digital audio watermarking
Psychoacoustic correction
ITU–R BS.1116–1

ABSTRACT

The article describes a new method of embedding and extracting the digital watermark in an acoustic signal, which operates in the frequency domain and uses a drift correction modulator of the watermark's phase angle, as well as a precise correction procedure of the watermark signal to the minimum masking threshold. The Drift Correction Modulation (DCM) has been worked out for the needs of the watermarking system with a large data payload and robustness against degrading factors which occur during signal processing. The digital watermark embedded in the original signal using the DCM method is inaudible and robust against the majority of degrading factors. The DCM method may be used in Digital Signal Processing applications, e.g. for a hidden authentication of subscribers in telecommunication channels or for determining the output phase mistuning of devices. DCM is characterized by a high normalized data payload of the watermark signal and may be implemented in applications of hidden and secure data transmission. The paper presents the results of auditory tests according to the ITU-R 1116-1 standard; moreover, results of the watermark extraction tests in a VHF radio link, with the use of military tactical radio stations have been presented.

1. Introduction

The Drift Correction Method (DCM) makes it possible to embed an additional signal containing a pre-set data payload in the host signal at the transmitter side of a telecommunications link and extract the signal, including the additional information, at the receiver side. The additional information is hidden in a phase angle monotonic change for selected additional signal harmonics.

The method of audio signal watermarking has been developed extensively since the 1990s [1–14], i.e. for the need of Copyrighting and Copy Distribution systems, and monitoring radio station listening figures. The basic repository for methods used in digital watermarking is the paper by Cox [15]. This book provides a list of methods of embedding additional signals in both time and transform domains.

The problem of estimating and correcting the frequency drift occurs in terms of bandwidth modulators used in telecommunications, including OFDM modulators. It is worth mentioning the paper by Moose [16] that describes the impact of frequency drift on digital communications and the effect of interference correction on many OFDM subcarriers. The paper by Moose describes the technique of frequency offset estimation using repeated data symbols aimed at determining the estimation error using the algorithm of Maximum Likelihood Estimation (MLE). The paper by Li et al. [17] describes an effective method for frequency drift in Wireless Local Area Networks (WLANs) based on the OFDM technique and using the predefined packet preamble structure. On the other hand, the paper by Shi and Serpedin [18] presents a problem of frequency drift estimation in reference to the sampling frequency of single subcarrier and multi-carrier modulated signals using modified Cramer–Rao lower bounds. A paper by Sliskovic [19] presents a method for frequency drift estimation and its correction, containing repeated data symbols and comparing the signal phase between successive repeated symbols on all subcarriers.

* Tel.: +48 22501334151; fax: +48 226839038.
E-mail address: Zbigniew.Piotrowski@wat.edu.pl.

0895-7177/$ – see front matter © 2011 Elsevier Ltd. All rights reserved.
doi:10.1016/j.mcm.2011.09.016

2. Problem formation

The determination of phase angle correction Δ (rad/s) is necessary if the watermarked in a frequency domain signal is transmitted in a telecommunications channel. Then, as a result of various and usually low stability quartz generators connected to D/A converters (transmitters) and A/D (receivers), the resultant frequency offset at the receiver side can be seen as a mono-directional angle phase drift Δ (rad/s) for all watermarked audio signal harmonics. The watermark embedder transmits the phase angle drift for selected watermark harmonics $X(m_p)$, referred to as "pilot signals" with zero value (without phase drift)

$$\Delta\chi_{\text{coder}} = 0 \text{ (rad/s)} \quad \text{for } X(m_p) \tag{1}$$

considering this, at the telecommunications channel receiving side we should expect exactly the phase angle value increment of $\Delta\chi = 0$ (rad/s). It is obvious that the telecommunications channel features a pre-set resultant offset $\Delta\phi_{\text{channel}} \neq 0$, so the resultant frequency offset for pilot signal harmonics m_p is:

$$\Delta\chi_{\text{out}} = \Delta\chi_{\text{coder}} + \Delta\chi_{\text{channel}} \text{ (rad/s)}. \tag{2}$$

Thus when $\Delta\chi_{\text{coder}} = 0$, the result is as follows:

$$\Delta\chi_{\text{out}} = \Delta\chi_{\text{channel}} \text{ (rad/s)}. \tag{3}$$

The offset can be determined using the phase angle scanning procedure described in the section regarding the DCM extractor.

In contrast to the methods described in [16,15,17–19], the problem of frequency drift of the additional signal embedded in the host signal is extremely complex. Considering the additional signal (degraded by the host signal) the value of the Signal-to-Mask ratio (SMR) resulting from the psychoacoustic correction procedure, where the so-called Just-Noticeable Difference level (JND) for the watermark signal is determined, the average values are significantly lower than −20 dB, which signifies, on the one hand, that the additional signal (watermark) is inaudible, whereas, on the other hand, it creates the problem of proper extraction of the additional signal and the information it carries. The paper describes a solution for watermark signal extraction as well as for phase angle drift estimation of the signal by using the phase angle scanning procedure.

3. Normalized data payload of the watermark signal

Each watermarking system using the technique of embedding a watermark robust against degrading factors may be characterized by a P_N parameter—a normalized data payload of the watermark signal. Thus,

$$P_N = \frac{P_{WM}}{T_{WM}} \text{ (bps)} \tag{4}$$

where: P_{WM}—the data payload of the watermark (b), T_{WM}—the duration of the signal on the receiving side (s), in which the watermark with a fixed data payload is decoded for fixed BER and JND.

The P_N parameter characterizes, in a simple way, any watermarking system with regard to the efficiency of decoding additional data and differentiates the worked out watermarking methods with regard to the data efficiency of the watermark. In watermarking systems, the duration of the decoded signal conditions the decoding of *complete* additional data. Thus, information about the watermark's data payload for a given method is not sufficient to characterize the information efficiency of the watermarking system. To differentiate, for hidden communication systems (steganographic) one of the system's efficiency measures is the bit rate (bps) which determines the amount of additional information sent within a time unit. For the same watermarking method, P_N will be different for various degrading factors; especially for the lack of signal degrading factors, the data payload should possess a maximum value marked as P_{NMAX}

In systems which provide a hidden and secure data transmission at the same time, the normalized data payload of the P_N watermark should be as high as possible, with a simultaneous fulfillment of the watermark's perceptual transparency condition and high robustness against basic degrading factors occurring e.g. in a telecommunications link. The use of the DCS method enabled us to obtain a data payload of the watermark signal equal to 168 bits and a normalized data payload of $P_{NMAX} = 5.6$ (bps) (BER=0%, Zero JND). This means that without degrading factors, e.g. lossy compression, AWGN noise, all-pass filtration, etc., during 30 s of the signal's duration complete additional information for BER $= 0$ (%) will be achieved. It is worth noting that P_N depends also on the type of original signal in which the watermark is embedded.

4. DCM coder

The diagram of the DCM coder has been presented in Fig. 1.

The binary signature is formed in the OFDM (Orthogonal Frequency Division Multiplexing) block to a series of harmonics with a randomly selected initial value of the phase angle φ_m, where m denotes the value of the subsequent spectral line, i.e. each m-th harmonic generated in the OFDM block will be described as:

$$x_m(t) = A_m e^{j\varphi_m} e^{j2\pi f_m t} \tag{5}$$

where: $A_m = 1$ and $f_m = f_1 m$, for $m = 1, 2, 3, \ldots, M$.

Fig. 2 illustrates the composite of harmonics in the frequency and time domain, generated in the OFDM block, for the set x_m with 27 spectral lines.

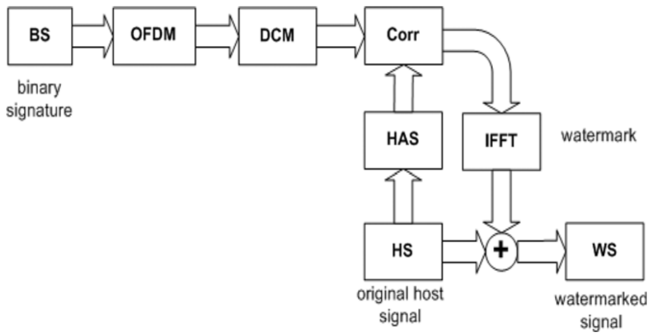


Fig. 1. The diagram of the DCM coder.

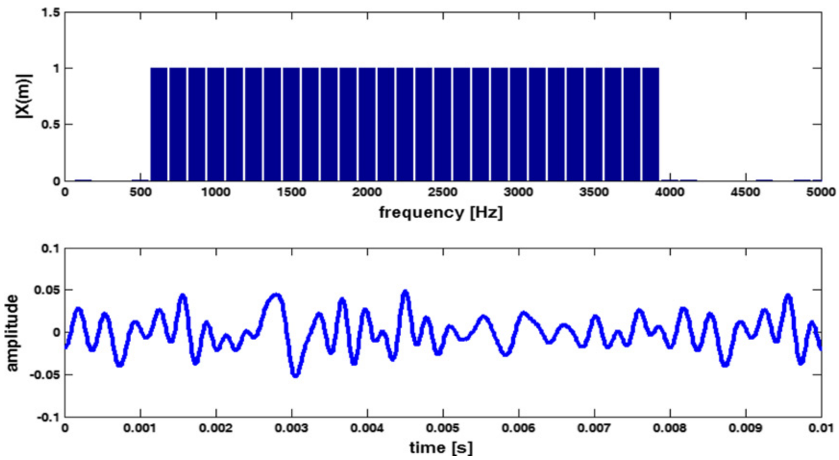


Fig. 2. Watermark signal formed in the OFDM block (illustration in the frequency and time domain).

In the described example, the first harmonic of the watermark signal starts from the 5th spectral line (the DC constant has an index of 0); thus, for 27 spectral lines, the last line lies in the 31 spectral bin $\langle 3750 - 3875 \rangle$ (Hz) at the frequency $f_{31} = 3812.5$ (Hz) (see Table 1). For system parameters: sampling frequency $f_s = 48$ kHz and FFT $= 384$ points, the designed pass band of each of the spectral bins equals 125 Hz. The watermark signal is included in the frequency band $\langle 562.5 - 3875.0 \rangle$ (Hz).

In the DCM block Fig. 2, an assigning of the binary signature occurs at the set constant increment value Δ of the phase angle χ_m (rad/s); thus, taking into account the formula (2) we may write down the end form of the m-th harmonic of the signal:

$$x_m(t) = A_m e^{j(2\pi f_m t + \varphi_m + \Delta\chi_m)}. \tag{6}$$

The procedure of setting the phase angle value χ_m is realized in the following steps:

1. The data payload of the P_{WM} watermark is declared. Harmonics with numbers $m_p = \{1, 5, 11, 17, 23, 27\}$, calculated from the first line in the watermark signal, fulfill the role of so-called pilots with a phase angle drift value $\Delta\chi_{\{1,5,11,17,23,27\}} = 0$ (rad/s) and no data bits are assigned to them. Thus, taking into account an L-ary of the DCM system, for $L = 256$, each spectral line will represent a $P = \log_2(256) = 8$ (b) information, meaning $P_{WM} = 168$ (b) for 21 spectral lines.
2. A unitary increment of the $\Delta\chi_1$ (rad/s) phase angle value is set for the first harmonic in the signal spectrum, calculated against the DC constant component, i.e. for system parameters: $L = 256$ and the last value of the watermark's spectral line for the 31 harmonic, we obtain:

$$\Delta\chi_{1\mathrm{deg/s}} = \frac{360}{31L} \text{ (deg/s)} \tag{7}$$

$$\Delta\chi_{1\mathrm{deg/s}} = \frac{\Delta\chi_{1\mathrm{deg/s}} 2\pi}{360} \text{ (rad/s)}. \tag{8}$$

3. Half of the phase angle's quantizer bin value is set:

$$K = \frac{2^P}{2}. \tag{9}$$

Table 1
Allocation of harmonics in the frequency spectrum for the DCM watermarking.

Spectral line no.	fs = 44, 100 Hz	Bit no.
	114.84	
	229.69	
	344.53	
	459.38	
1	574.22	Pilot 1/6
2	689.06	Info 1
3	803.91	Info 2
4	918.75	Info 3
5	1033.59	Pilot 2/6
6	1148.44	Info 4
7	1263.28	Info 5
8	1378.13	Info 6
9	1492.97	Info 7
10	1607.81	Info 8
11	1722.66	Pilot 3/6
12	1837.50	Info 9
13	1952.34	Info 10
14	2067.19	Info 11
15	2182.03	Info 12
16	2296.88	Info 13
17	2411.72	Pilot 4/6
16	2526.56	Info 14
19	2641.41	Info 15
20	2756.25	Info 16
21	2871.09	Info 17
22	2985.94	Info 18
23	3100.78	Pilot 5/6
24	3215.63	Info 19
25	3330.47	Info 20
26	3445.31	Info 21
27	3560.16	Pilot 6/6

4. Specific C_{dec} decimal values are assigned to the $\Delta\chi_m$ phase angle drift value for the m-th harmonic ($\Delta\chi_m = \Delta\chi_1 m$). The decimal C_{dec} values are calculated based on the binary signature value for blocks in the number of 8 (b). For example, for $P_{WM} = 168$ (b) we have 21 blocks, each containing 8 bits, and each of the 21 blocks is converted to a decimal form. The allocation of the $\Delta\chi_m$ value occurs according to the procedure written down in the following pseudocode A1:

for $i = 1$ to P_{WM} **do**
  if $C_{dec} \leq K$ **then**
    $\triangle\chi_m \leftarrow (\triangle\chi_1 C_{dec})(-1)$
  else
    $\triangle\chi_m \leftarrow (\triangle\chi_1 C_{dec}) - K$
  end if
  $i \leftarrow i + P$
end for

The model of the Human Auditory System (HAS) has been performed using mathematic methods taking advantage of the so-called psychoacoustic analysis i.e. to determine the value of minimum masking threshold ($LT_{\min}$) as well as the real spectral image perceived by the human auditory system as sound stimuli with no irrelevant and redundant components.

In the Corr block Fig. 1, a precise watermark signal shaping is performed to the minimum masking threshold level ($LT_{\min}$), with the use of a psychoacoustic HAS model described in the standard [20] for the MPEG standard in layer 1. The procedure described in [21] realizes a watermark signal shaping at one stage to the minimum masking threshold level based on the determined shaping coefficient for each of the analyzed 32 frequency sub-bands in each signal frame. Each of the corrected signal frames contains 384 samples. The determined shaping coefficient values coef_{WM} are written down as:

$$\mathrm{coef}_{WM} = \mathrm{SPL}_{WM} + (\mathrm{Delta}_H - \mathrm{Delta}_{WM}) - LT_{\min} + C \tag{10}$$

where: coef_{WM}—shaping coefficient values (dB) for the watermark signal, SPL_{WM}—the Sound Pressure Level (dB) for the watermark signal, Delta_H normalization coefficient for the original signal, Delta_{WM}—normalization coefficient for the watermark signal with reference to 96 dB SPL, C (optional)—constant value added in order to determine the SMR (dB).

By correcting the watermark signal to the level of minimum masking threshold ($LT_{\min}$), it is possible to provide the so-called Just-Noticeable Difference (JND) level for which listeners in the 50% of auditory trials cannot recognize the difference

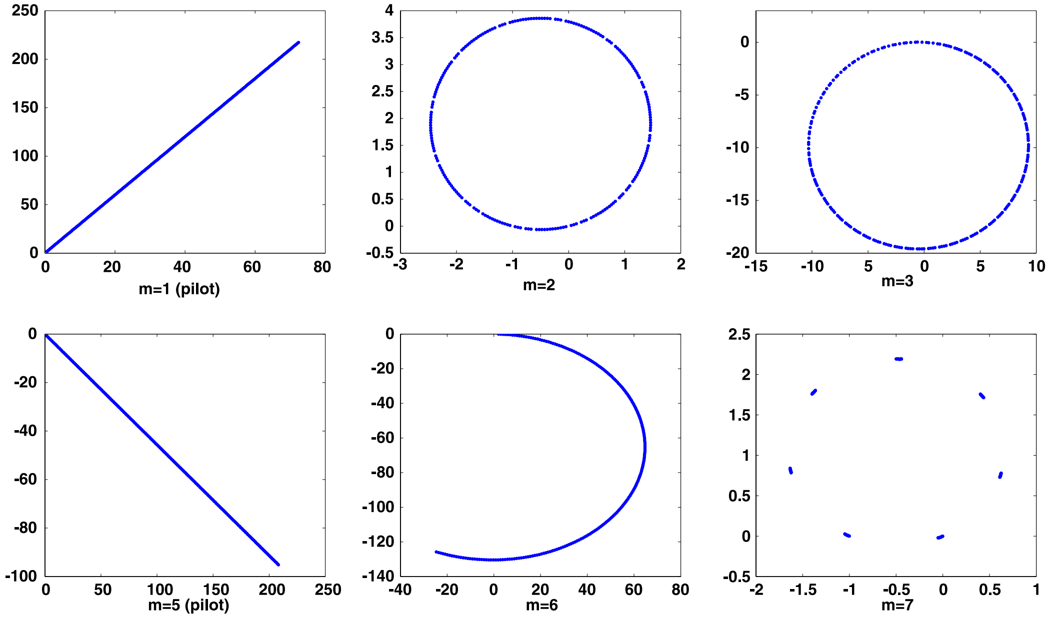


Fig. 3. Harmonics of the watermark signal, $m = \{1, 2, 3, 5, 6, 7\}$ illustrated on planes (Im Re) for $N = 250$ of signal frames (the frame consists of 384 samples).

between the watermarked and host signal. The JND level defines the potential perceptual transparency of the watermark embedded within the host signal. The JND level depends on the signal-to-mask ratio (SMR) (dB) that defines the difference between the host signal and watermarking signal.

In the Corr block, after correcting the watermark signal to the JND level, a phase angle correction of each of the watermark's spectral lines is also performed. The correction is performed for each signal frame with a monotonously increasing or decreasing (depending on the sign) increment value $\Delta\chi_m$. The $\Delta\chi_m$ values for each of the watermark's spectral lines are determined in the DCM block. Thus, the increment of the phase angle value $\varDelta$ for a selected m-th of the spectral line $X(m)$ representing the watermark for two subsequent signal frames equals exactly $\Delta\chi_m$

$$\Delta\chi_m = X_{\Phi 1}(m) - X_{\Phi 2}(m) \tag{11}$$

where: $X_\Phi(m) = \operatorname{arc\,tg}\left(\frac{X_{\text{imag}}(m)}{X_{\text{real}}(m)}\right)$.

In addition, two conditions should be fulfilled during the phase angle drift correction for each of the watermark's harmonics: 1. The value $\Delta\chi_m$ for the determined m-th number of the harmonic, calculated against the DC constant component, must be incremented by a successive index of the signal frame k_{var} for the determined index of the harmonic m_{const}:

$$\Delta\chi_m = m_{\text{const}}\chi_m k_{\text{var}} \tag{12}$$

thus, for the m of this line, k_{var} (where k is the subsequent index of the signal frame).

2. In the case when for a given signal frame the phase angle value fulfills the inequality:

$$\Delta\chi_m > 2\pi, \quad \text{or} \quad \Delta\chi_m \leq 2\pi \tag{13}$$

a correction should be performed of the phase angle value $\Delta\chi_m$ by 2π.

Based on the original signal, a minimum masking threshold $LT_{\min}$ is set in the HAS block Fig. 1. This threshold is necessary to correct the watermark signal in the Corr block Fig. 1. In the IFFT block Fig. 1, a transformation of the watermark occurs from the frequency domain to the time domain, while in the adder block the original host signal is added to the watermark signal. Thus, on the output of the DCM coder we are dealing with an original signal that is marked with a watermark. The watermark signal is inaudible in the presence of a host signal. The watermark signal has monotonous phase angle increments (constant increments $\Delta\chi_m$ (rad/s)) determined for each harmonic of the watermark signal.

Fig. 3 illustrate the phase angle drift (constant increments $\Delta\chi_m$ (rad/s)) for six selected harmonics of the watermark signal $m = \{5, 6, 8, 10, 11, 14\}$ that have been illustrated on planes (Im Re). A clear lack of phase angle drift for harmonics with numbers $m = 1$ and $m = 5(\Delta\chi_{(1,5)} = 0$ (rad/s)).

Fig. 4 illustrates, in the time and frequency domains, the original host signal and the watermark signal obtained as a result of the DCM coder's signal processing.

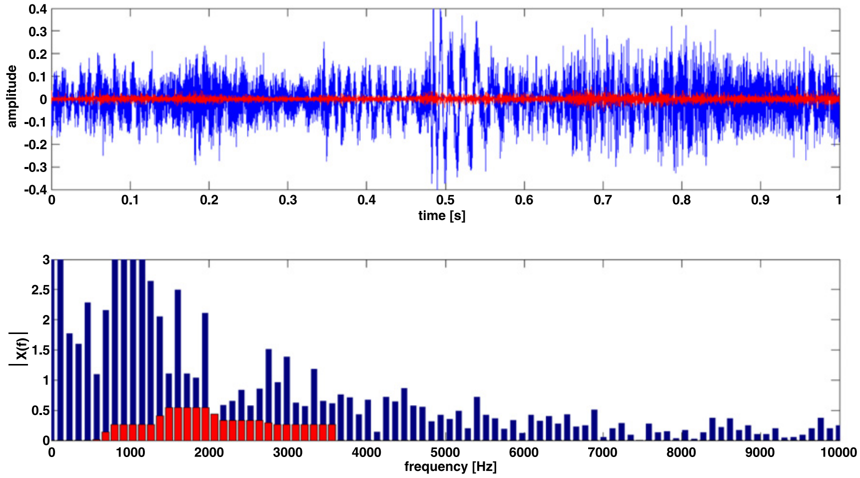


Fig. 4. Original signal and watermark signal (red color) in the time and frequency domain.

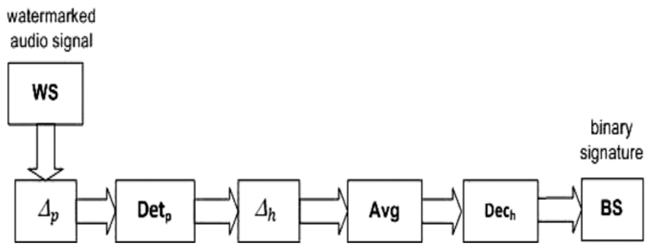


Fig. 5. DCS decoder diagram.

5. DCM extractor

The diagram of the watermark decoder has been presented in Fig. 5.

The DCM decoder belongs to a class of blind-type systems and does not require knowledge about the original signal in the watermark extraction procedure. The audio signal with an embedded watermark is transferred to the block Δ_p (Fig. 5). In this block, the increment Δ (rad/s) of the phase angle for the spectral lines of pilots $X(m_p)$ is computed, where $m_p = \{5, 6, 8, 10, 11, 14\}$, and the determined values constitute a correction to the phase angle shaping of all spectral lines for the audio signal with an embedded watermark.

The mistuning $\Delta\chi_{\text{out}}$ is found in the phase angle scanning procedure (phase angle scanner) described thoroughly in the article [22]. This procedure includes setting a maximum value of the virtual harmonic module written down as:

$$X_V = \sum_{m=1}^{M} (|X_{m\chi}|) \tag{14}$$

where: X_V—virtual harmonic, $X_{m\chi}$—complex value of the m-harmonic of the pilot signal for a determined phase angle reduction coefficient value χ. The virtual harmonic is formed from six pilot signal harmonics (m_p), $M = 6$. Thus, we write that:

$$X_{m\chi} = \sum_{k=1}^{K} \text{Re}(X_{mk}) + \sum_{k=1}^{K} \text{Im}(X_{mk}). \tag{15}$$

The virtual harmonic $X_{m\chi}$ is set by averaging spectral line values of the same m index in K iterations (K—the number of signal frames). The FFT complex values are averaged separately for the real and imaginary parts, with a set value χ of the phase angle reduction coefficient in K iterations. We compute a maximum value of the virtual spectral line module for a determined value χ of the phase angle reduction coefficient

$$X_{v\max} = \max \sum_{m=1}^{M} (|X_{m\chi}|). \tag{16}$$

When averaging the spectrum of a watermark signal, the value of the phase angle reduction coefficient is selected from the range $\chi_{\text{range}} \in \{-\chi_{\min}, \chi_{\max}\}$. In the case of obtaining a maximum value $X_{V\max}$ of the module, calculated for the i-th

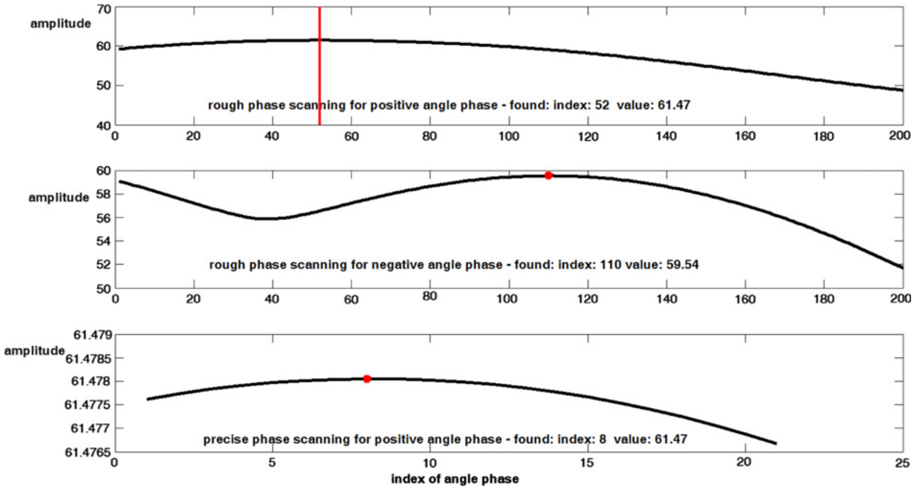


Fig. 6. Illustration of the phase angle scanner processing, found correction of the phase angle $\chi = -7, 19 \cdot 10^{-4}$ (rad/s).

value χ_i, selected from the range χ_{range}, the value χ_i (rad/s) is specified as the phase angle correction which reduces the output phase angle drift $\Delta\varphi \to 0$ (rad/s). In the Δ_p block the signal with an embedded watermark is shaped, based on the calculated correction. Thus, taking into consideration Eq. (5) we may write down the signal after the phase angle correction as:

$$x_m(t) = A_m e^{j(2\pi f_m t + \varphi_m + \chi_{\text{out}} - \chi_i)} = A_m e^{j(2\pi f_m t + \varphi_m)}. \tag{17}$$

The result of searching for the phase angle correction with the use of a phase angle scanner has been presented in Fig. 6.

Through averaging the signal spectrum with the watermark with the fulfilled condition $\Delta\varphi \to 0$, we obtain a reduction in the spectrum of those components which do not repeat themselves periodically (original host signal), and we strengthen the components of the watermark signal. Thus, we may write down the gain of the coherent averaging as:

$$SNR_{\text{coh}}(\text{dB}) = 20\log_{10}(SNR_{\text{coh}}) = 20\log_{10}(\sqrt{K}) = 10\log_{10}(K). \tag{18}$$

The value of the SNR coefficient depends on the number of K iterations (the number of averaging for subsequent signal frames). If, e.g. SMR = 25 dB, i.e. the watermark signal is "hidden" below 25 dB from the original host signal level, in order to correctly decode the watermark, we would need a number of K averaging of the signal frames equal to:

$$K = 10^{\frac{25}{10}} = 316. \tag{19}$$

In the Det_p block the detection of pilots occurs, in the case of detecting spectral lines with a drift value equal to $\Delta_\varphi = 0$ (rad/s). The spectral lines are detected already after a correction of the output mistuning with a computed reduction coefficient Δ_φ in block Δ_p. If the number of detected pilots is higher than or equal to 4, it is assumed that the audio signal contains a watermark. Thus, pilot lines fulfill two functions:

1. They are a reference signal for calculating the phase angle shaping for an output mistuning,

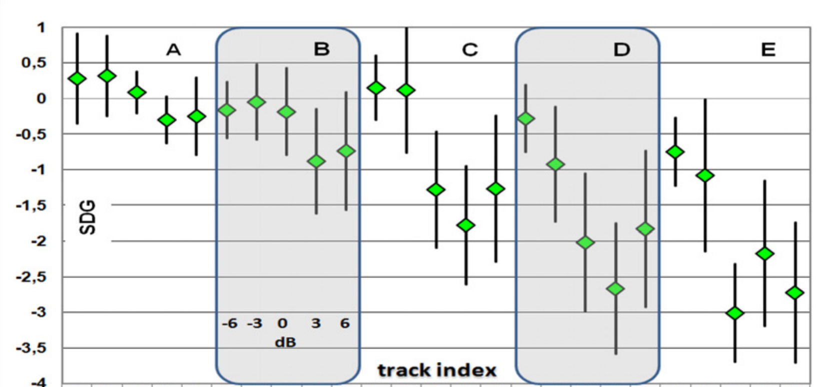
$$\Delta\chi_{\text{out}} = \Delta\chi_{\text{channel}} \text{ (rad/s)}. \tag{20}$$

2. They play the role of a preamble protecting against false-positive errors.

In the block marked on Fig. 5 as Δ_h, the drift of the phase angle is determined for each harmonic of the watermark, with the exclusion of harmonic pilots $X(m_p)$. This operation uses the phase angle scanning procedure (described for block Δ_p). The computed values $(\Delta\chi_m)$ (rad/s) are exactly those that decided about the phase angle drift for harmonics in the DCS coder. In the Dec_h block the decoding of the binary signature occurs based on the allocation of value $\Delta\chi_m$ for specific binary values based on pseudocode A1. As a result of the DCS decoder's activity, the binary signature of the watermark is obtained.

The initial angle phase $\varphi_m(3)$, for each harmonic, is also possible to detect using a DCM extractor. Then, in this scheme, we can obtain an additional data payload. The DCM extractor read the cumulated angle phase value for each harmonic (21):

$$\varphi_m = \arctan \frac{\sum_{k=1}^{K} \text{Im}(\chi_m)}{\sum_{k=1}^{K} \text{Re}(\chi_m)}. \tag{21}$$



	1	2	3	4	5	6	7	8	9	10	11	12	13	14	15	16	17	18	19	20	21	22	23	24	25
Max	0,92	0,89	0,38	0,03	0,29	0,23	0,48	0,43	-0,1	0,09	0,61	1,01	-0,5	-1	-0,2	0,19	-0,1	-1	-1,8	-0,7	-0,3	-0	-2,3	-1,2	-1,7
Min	-0,4	-0,2	-0,2	-0,6	-0,8	-0,6	-0,6	-0,8	-1,6	-1,6	-0,3	-0,8	-2,1	-2,6	-2,3	-0,8	-1,7	-3	-3,6	-2,9	-1,2	-2,1	-3,7	-3,2	-3,7
Mean	0,28	0,32	0,09	-0,3	-0,3	-0,2	-0,1	-0,2	-0,9	-0,7	0,15	0,12	-1,3	-1,8	-1,3	-0,3	-0,9	-2	-2,7	-1,8	-0,8	-1,1	-3	-2,2	-2,7

Fig. 7. ITU-R BS—1116.1 subjective audio fidelity test result based on [23].

6. Tests

In order to execute the procedure of authorizing a subscriber in telecommunication channels, one of the known authorization models was selected which involves sending a recognizable binary signature to subscribers who exchange correspondence with each other. However, it must be stressed that the authorization model has been supplemented with hidden signature sent with the use of a watermark signal added to the conversation signal, which constitutes its considerable modification. This section presents the results of experiments related to authorizing a subscriber using the technique of information hiding in the Very High Frequency (VHF) radio channel. For the tests related to authorizing a subscriber in the radio channel, the DCM method was proposed. In order to test the robustness of this method against degrading factors, including those in the VHF radio channel (30 MHz–90 MHz), several tests were carried out, whose results are presented below.

Experiments were conducted regarding:

- the estimation of the perceptual transparency of an acoustic signal with an embedded watermark signal,
- the effectiveness of decoding the original signal without an embedded watermark (robustness against false-positive error types),
- the effectiveness of decoding the original signal with an embedded watermark,
- the robustness of the watermark against lossy compression,
- the robustness of the watermark against transmission in the VHF link.

6.1. Estimation of perceptual transparency

The standard [20] contains an exact description of measuring the degraded sound's quality and the acoustic conditions which have to be fulfilled during carrying out acoustic tests. The test described by the BS.1116-1 [23] norm is a standard for examining signals subjected to marking. Parameters: 16 listeners with normal hearing, 6 audio tracks: A—pop music, B—classical Vivaldi, C—English male speech, D—German male speech, E—viola. According to the definition of the Just Noticeable Difference given in [24], JND is a level of distortion that can be perceived in 50% of experimental trials. Each audio track was presented with various masking threshold LT, where: $LT = LT_{\min} + G$ and $G = \{-6, -3, 0, 3, 6\}$ dB. For JND $= 0$ we have masking threshold $LT = LT_{\min}$ computed for the original, host signal. The results are computed for a 95% confidence interval. The results of the BS 1116–1 test have been presented in Fig. 7.

The watermark signal embedded in the original signal through the DCS coder is perceptually transparent (inaudible). The listeners assessed that the watermarked signal is imperceptible to them (Subjective Diff-Grades (SDG) > 1.1), and cases have even been reported of an erroneous differentiation between the degraded and original signal (SDG values above zero). The introduction of a statistical analysis decreases the spread of estimated values around the average value (for 95% of the confidence interval). The Forced Choice (percent of correct answers) is shown in Fig. 8.

6.2. Off-line decoding efficiency

In off-line mode test (based only on a PC as watermark embedder and extractor), the digital signal was decoded with no telecommunication channel, taking into account 5 watermarked audio tracks. The signal metrics have been detailed in Table 2. Legend: 001—pop music, 002—classical Vivaldi, 003—English male speech, 004—German male speech, 005—viola. We denote: P_{host}—power of the host signal in (dB), P_{zwcorr}—power of the watermark (dB), P_{wm}—power of the watermarked

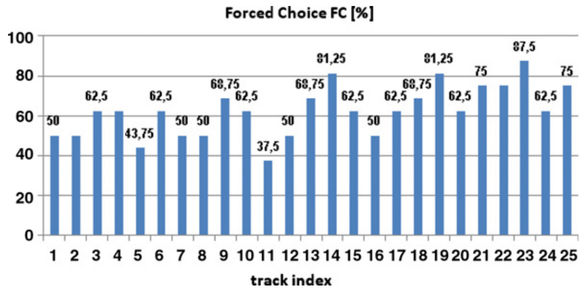


Fig. 8. The FC (Forced Choice) coefficient vs. track index.

Table 2
Audio tracks metrics, computed for 30 (s) length of each track.

Track no.	Phost	Pzwcorr	Pwm	SMR	MSE	NMSE	PSNR	AF
1	45.2573	26.0387	45.3120	19.2186	−38.5552	−19.2186	−28.2285	0.988
2	34.2896	13.7526	34.3251	20.5370	−50.8413	−20.537	−27.6117	0.9912
3	44.3821	21.4372	44.4068	22.9448	−43.1567	−22.9448	−25.743	0.9949
4	41.8365	20.8766	41.8745	20.9599	−43.7173	−20.9599	−24.549	0.992
5	47.7259	22.9712	47.7386	24.7546	−41.6227	−24.7546	−28.5955	0.9967
Mean	42.698	21.015	42.731	21.683	−43.579	−21.683	−26.946	0.993
Min	34.29	13.753	34.325	19.219	−50.841	−24.755	−28.596	0.988
Max	47.726	26.039	47.739	24.755	−38.555	−19.219	−24.549	0.997
Std	5.15	4.528	5.146	2.176	4.528	2.176	1.732	0.003

signal (dB), *SMR Signal-to-Mask Ratio* (dB), MSE *Mean Square Error* (dB), NMSE *Normalized MSE* (dB), PSNR *Peak Signal to Noise Ratio* (dB), AF *Audio Fidelity*.

Parameters: 8 bits per harmonic, 21 harmonics, payload $P = 168$ (b). Watermark extracting process: audio track length 30 (s), frequency of sampling fs = 48 kHz, for $LT = LT_{\min}$

Results: for each audio track we received BER = 0 (%), thus according to the formula (1), we write that: $P_{NMAX} = 5.6$ (b) (normalized payload), BER = 0%.

6.3. False-positive type errors

The test took into account 5 audio signal tracks without an embedded watermark signal. The signal metrics have been detailed in Table 2. The watermark extracting process: various audio track lengths {1, 10, 20, 30} (s), frequency of sampling fs = 48 kHz.

Results: the watermark binary signature was not detected in all tracks. The 6 pilot harmonics were not detected with the angle phase drift equal to formula (1).

6.4. Decoding efficiency vs data payload

The test took into account 1 track (track signature no.: 001) with an embedded watermark. The signal metrics have been detailed in Table 2. Parameters: 21 harmonics, $P = \{21, 42, 63, 84, 105, 126, 147, 168, 189, 210, 231, 252, 273, 294, 315\}$ (b) (payload) audio track length: 20 s, fs = 48 kHz, for $LT = LT_{\min} + 3$ dB$P_N = \{1.05, 2.1, 3.15, 4.2, 5.25, 6.3, 7.35, 8.4, 9.45, 10.5, 11.55, 13.65, 14.7, 15.75\}$

6.5. Resampling (off-line embedding and extracting)

The test took into account 5 audio signal tracks (no. 001–005) with an embedded watermark The signal metrics have been detailed in Table 2.

Parameters: payload $P = 168$ (b), 8 bits per harmonic, 21 harmonics, audio track length: 30 s, fs = 48 kHz, for $LT = LT_{\min} + 3$ dB. Resampling scheme: *.wav (fs = 48 kHz, 16 b/sample) →*.wav (fs = 8 kHz, 16 b/sample) →*.wav (fs = 48 kHz, 16 b/sample)

Results: for each resampled and decoded audio tracks, received BER = 0 (%). We can write the value of the normalized payload: $P_N = 5.6$ (bps), BER = 0 (%).

6.6. MP3 compression vs. data payload

The test took into account 1 audio signal track (no. 001) with an embedded watermark. The signal metrics have been detailed in Table 2.

Parameters: payload $P = \{21, 42, 63, 84, 105, 126, 147, 168, 189, 210, 231, 252, 273, 294, 315\}$ (b) audio track length: 30 s, fs $= 48$ kHz, for $LT = LT\min + 3\text{dBP}_N = \{0.7, 1.4, 2.1, 2.8, 3.5, 4.2, 4.9, 5.6, 6.3, 7.0, 7.7, 9.1, 9.8, 10.5\}$ (bps) MP3 lossy compression with compression rate: 6:1, 128 kbps. Scheme: *.wav (fs = 48 kHz, 16 b/sample) →*.mp3 (fs = 48 kHz, 6 : 1) →*.wav (fs = 48 kHz, 16 b/sample).

Results: for compressed track with dedicated data payload, received BER = 0 (%), thus mp3 compression with compression rate 6:1 has no effect on bit errors occurring during the decoding process.

6.7. MP3 compression vs. compression rate

The test took into account 1 audio signal track (no. 001) with an embedded watermark. The signal metrics have been detailed in Table 2.

Parameters: payload $P = 105$ (b), 5 bits per harmonic, 21 harmonics, audio track length: 30 s, fs $= 48$ kHz, for $LT = LT_{\min} + 3\ \text{dB}P_N = 3.5$ (bps) MP3 lossy compression with compression rates: 12:1 (64 kbps), 9.6:1 (80 kbps), 8:1 (96 kbps), 6.9:1 (112 kbps), 6:1 (128 kbps). Scheme: *.wav (fs = 48 kHz, 16 b/sample) →*.mp3 (fs = 48 kHz, variablecompressionrates) →*.wav (fs = 48 kHz, 16 b/sample).

Results: for compressed track with dedicated compression rate, received BER = 0 (%), thus mp3 compression rates from: 6:1 up to 12:1 have no effect on bit errors occurring during the decoding process.

6.8. VHF radio link

Aim: Examining the robustness of the watermark embedded in the acoustic signal against transmission in the VHF channel. Description of experiment: a signal with a watermark embedded through the DCM coder was subjected to extracting. The watermarked signal was then transmitted through an VHF radio link with the use of military radio station Radmor TRC 9200. Transmission parameters: transmitting and receiving frequency 38 MHz (simplex mode), F3E modulation in the HLG radio station work mode at a set frequency. The test took into account 1 audio signal track (no. 001) with an embedded watermark. The signal metrics have been detailed in Table 2.

Parameters: payload $P = \{21, 42, 63, 84, 105, 126, 147, 168, 189, 210, 231, 252, 273, 294, 315\}$ (b) audio track length: 30 s, fs $= 48$ kHz, for $LT = LT_{\min} + 3\ \text{dB}P_N = \{0.7, 1.4, 2.1, 2.8, 3.5, 4.2, 4.9, 5.6, 6.3, 7.0, 7.7, 9.1, 9.8, 10.5\}$ (bps) VHF radio link, HLG mode with fixed frequency 38 MHz, military TRC 9200 radio station. Scheme: *.wav (fs = 48 kHz, 16 b/sample) → D/A → VHF radio link → A/D →*.wav (fs = 48 kHz, 16 b/sample).

Results: for track with watermarked data payloads {63, 84, 105, 126, 147} received BER = 0 (%). For other data payloads the watermark extractor did not detect the watermark (number of detected pilots <4). Remarks:

- It is possible to transmit a watermark embedded in an acoustic signal through a DCM coder in a VHF radio channel. The obtained WM binary payload is 147 (b) for BER = (0%) and it points out the possibility of embedding the watermark's binary signature during a hidden transmission in the VHF radio channel.
- The hidden binary signature sent through the VHF radio channel constituted a Personal Identification Number (PIN), through which it is possible to verify the identity of the subscriber on the receiving side.
- An undeniable, significant downside to the developed and tested DCM system operating on radio channels is the condition imposed for the duration of the signal transmitted in a continuous way (the simplex mode switch of the radio station turned off) which equals minimum 30 (s), for an average WM signal power equal to $LT\min + 3$ dB. The continuous transmission time may be decreased at the cost of increasing the average WM signal power.

7. System comparison

The DCM method features a normalized data payload of binary watermark structure of 5.6 (bps) at BER = 0 (%). For comparison, the normalized data payload of professional, commercial audio signal watermarking systems ranges from 2.4 (bps) (Verance, DVD-A/SDMI-1 system), 1.4 (bps) (Philips Research, CompoTrackWAV system) to 18.8 (bps) (Eric Mtois, Eym Audio Watermark system) and 18 (bps) (Alex Radzishevsky: Audio Watermarking Tools system).

8. Conclusions

The proposed DCS scheme allows for obtaining a high normalized data payload P_{NMAX} of the watermark signal embedded into an audio signal. In the watermark coder new technologies have been used: phase angle drift correction based on the DCM watermark modulator, psychoacoustic correction based on a single-stage process of the watermark signal shaping to the JND level. In the extractor, the procedures of phase angle scanning and coherent signal spectrum averaging have been used. As a result, the implemented DCS system generates an audio signal marked with a watermark which is inaudible in the presence of an original, host signal and is robust against degrading factors: lossy compression, resampling, A/D and D/A conversion, VHF channel noise. The conducted tests proved that the audio signal with a watermark embedded according to the described method may be used for marking a UKF radio transmission. Because of the high value of the coefficient P_{NMAX}, the system may potentially be used in applications for hidden and secure data transmission in a radio link [25], e.g. with the use of a Personal Trusted Terminal. The phase angle drift scanner method may, in turn, be used in a precise estimation of the frequency mistuning in measurement systems and, among other things, for reproducing synchronization in telecommunication devices. The described DCM method is also dedicated for radio communications systems to establish hidden channels or additional service channels based on existing OFDM modulators e.g. in systems: IEEE 802.11, DVB-T, 4G etc. Then, the input audio host signal is replaced, in the described DCM system, with an OFDM high frequency signal and an additional OFDM signal is embedded as a digital watermark by the DCM embedder.

References

[1] M.S. Swanson, et al., Robust audio watermarking using perceptual masking, Signal Processing 66 (3) (1998) 337–355.
[2] P. Bassia, I. Pitas, Robust audio watermarking in time domain, in: Proc. of the 9th European Signal Processing Conference, EUSIPCO-98, 8-11 September 1998.
[3] D. Gruhl, Echo Hiding, in: Ross Anderson (Ed.), Proc. of First Int. Workshop on Information Hiding, Cambridge, UK, May 30–June 1, in: Lecture Notes in Computer Science, vol. 1174, 1996, pp. 295–316.
[4] M. Arnold, Audio watermarking: features, applications and algorithms, in: IEEE Proceedings 2000, Department for Security Technology for Graphics and Communication Systems Fraunhofer—Institute for Computer Graphics.
[5] W. Bender, D. Gruhl, N. Morirrnoto, A. Lu, Techniques for data hiding, IBM Systems Journal 35 (3–4) (1996) 313–336.
[6] P. Bassia, I. Pitas, Robust audio watermarking in the time domain, signal processing IX, theories and applications, in: Proceedings of Eusipco-98, Ninth European Signal Processing Conference, Rhodes, Greece, 8–9 September 1998.
[7] Changsheng Xu, Jiankang Wu, Qibin Sun, Digital audio watermarking and its applications in multimedia database, in: Signal Processing and its Applications ISSPA99Brisbane, Australia, 22–25 August, 1999.
[8] Ch. Neubauer, Digital watermarks for uncompressed and compressed audio signals, in: Fraunchofer Institute for Integrated Circuits Workshop Sicherheit in Mediendaten September 19th 2000.
[9] B. Ko, R. Nishimura, Y. Suzuki, Time-spread echo method for digital audio watermarking using PN sequences, Proceedings of the IEEE (2000).
[10] F. Hartung, M. Kutter, Multimedia watermarking techniques, Proceedings of the IEEE 87 (7) (1999) 1079–1107.
[11] M. Swanson, M. Kobayashi, A. Tewfik, Multimedia data embedding and watermarking technologies, Proceedings of the IEEE 86 (6) (1998) 1064–1087.
[12] J.D. Grody, L.T. Bruton, Performance evaluation of digital audio watermarking algorithms, in: Proceedings 43rd IEEE Midwest Symposium on Circuits and Systems, Lansing, MI, August 8–11, 2000.
[13] N. Wirth, Algorithms and Data Structures, Prentice Hall, Englewood Cliffs, NJ, 1986.
[14] Hyen O. Oh, Jong Won Seok, Jin woo Hong, Dae Hee Youn, New echo embedding technique for robust and imperceptible audio watermarking, IEEE Proceedings (2000).
[15] I.J. Cox, M.L. Miller, J.A. Bloom, Digital Watermarking, Morgan Kaufmann Publishers, ISBN: 1-55860-714-5, 2002.
[16] P.H. Moose, A technique for orthogonal frequency division multiplexing frequency offset correction, IEEE Transactions on Communications 42 (10) (1994) 2908–2914.
[17] J. Li, G. Liu, G.B. Giannakis, Carrier frequency offset estimation for OFDM-based WLANs, IEEE Signal Processing Letters 8 (3) (2001) 80–82.
[18] K. Shi, E. Serpedin, On the estimation of sampling clock frequency offset in single and multiple carrier systems, IEEE Transactions on Communications (2005) 408–412.
[19] M. Sliskovic, Sampling frequency offset estimation and correction in OFDM systems, in: The 8th IEEE International Conference on Electronics, Circuits and Systems, vol. 1, 2001, pp. 437–440.
[20] ISO CD11172–3 Coding of moving pictures and associated audio for digital storage media at up to about 1.5Mbit/s Part 3 Audio.
[21] Piotrowski Zbigniew, Precise psychoacoustic correction method based on calculation of JND level, Acta Physica Polonica, A 116 (3) (2009) 375–379. Optical and Acoustical Methods in Science and Technology.
[22] Gajewski Piotr, Lopatka Jerzy, Piotrowski Zbigniew, A new method of frequency offset correction using coherent averaging, Journal of Telecommunications and Information Technology 1 (2005) National Institute of Telecommunications, Warsaw 2005. ISSN: 1509-4553.
[23] ITU–R BS.1116–1, Methods for the subjective assessment of small impairments in audio systems including multichannel sound systems. http://www.itu.int/rec/R-REC-BS.1116-1-199710-I/en.
[24] Ingemar J. Cox, Metthew L. Miller, Jeffrey A. Bloom, Digital Watermarking, Morgan Kaufmann Publishers, ISBN: 1-55860-714-5, 2002.
[25] Piotrowski Zbigniew, The national network-centric system and its components in the age of information warfare, in: Safety and Security Engineering III, WIT Press, ISBN: 978-1-84564-193-1, 2009, pp. 301–309.