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Stealthy Information Transmission in the Terrestrial GMDSS Radiotelephone Communication

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ABSTRACT: Audio watermarking (AW) technology is considered for stealthy information embedding directly into audio signal. Inherent in analog radiotelephone channel interferences against watermarks are analyzed. Robust encoding/decoding algorithms are presented and appropriate project of AW system named as Automatic Radio Telephone Identification System (ARTIS) is proposed. Experimental results for the practical VHF radio channel are presented. Relating on processing complexity the designed system enables imperceptibly transmit data on a rate up to 260 bit/sec in the standard VHF radio channel. ARTIS provides the full compatibility with the existing radio installation, and doesn't require replacement of standard VHF transceivers and operational procedures. Besides, automatic identification the system may be used in the special applications, for example, under the threat of terrorist attack; generally contributes to navigation safety and information security.

1 INTRODUCTION

Audio watermarking (AW) corresponds to digital information imperceptibly embedded into the audio signal. AW for maritime terrestrial radiotelephony is inspired first of all by the ability of implementation an automatic identification in the voice telephone communication. Applied to the existing analog radiotelephony in all frequency bands (VHF, MF, HF), the watermarking system could overcome existing limitations, enhance efficiency and clearness of radiotelephone messaging while ultimately decreasing so called "human factor".

In the maritime and aeronautical VHF services analogue channels with frequency/phase and amplitude modulation correspondingly are utilized.

For the meanwhile the identification of the sea vessels is realized by means of verbal calling of ship's call sign or numerical identification. However on account of different reasons such verbal identification may be absent, transmitted with delay, or understood with errors. This problem, applied to VHF communication is illustrated in Fig. 1. Motor vessel "Arcona" transmits a certain message to all stations. But one of the receiving vessels missed the name and call sign of the transmitting ship, and another ship interpreted the name of transmitting ship as "Gargona" instead "Arcona".

It is obvious that false, incorrectly interpreted or delayed verbal identification negatively affects maritime navigation. Automatic identification could avoid misidentification and call sign confusion.

Radio Regulation claims obligatory identification for every radio transmission. In a present maritime terrestrial radiotelephony identification is completely dependent upon operator. This situation takes place on the background of high technology instruments is being implemented in the maritime branch.

It is well known an Automatic Transmitter Identification System (ATIS) [1] for marine VHF radio that is used and mandated on inland waterways in Europe for identifying the transmitting vessel. In ATIS the identity of the vessel is sent digitally immediately after the ship's radio operator has finished talking and releases push-to-talk (PTT) button. Identification is performed by appending a short data message in Digital Selective Calling (DSC) format. The main drawback of ATIS is post-report transmission of identification data.

In COMSAR proposal [2] the necessity of automatic identification is grounded and quite reasonably noted that the identification should be done immediately after pressing the PTT button on the contrary of ATIS releasing PTT. Another shortcoming of ATIS is principle limitation for Medium/High frequency (MF/HF) applications. This limitation results from the small bitrate (100 bit/sec) in comparison to VHF DSC rate (1200 bit/sec).

In restricted navigation environment the immediate and clear automatic identification is extremely necessary. Automatic identification would exclude the human factor and increase an efficiency of VHF radiocommunication and maritime safety in the whole.

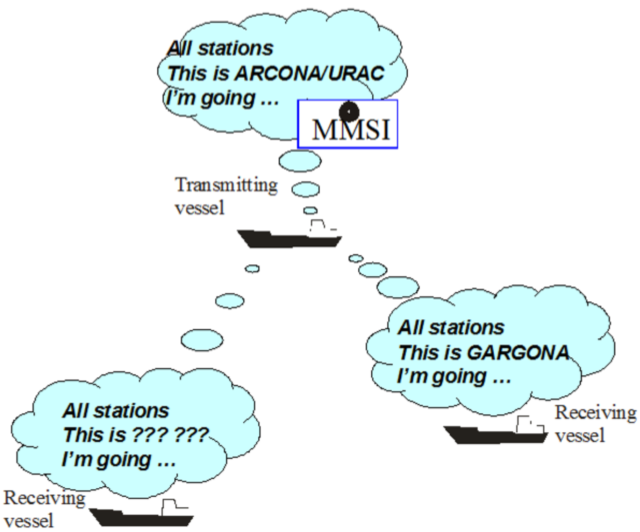

Figure 1. Automatic identification in the VHF maritime radiotelephony

Only verbal identification doesn't protect against illegal radio transmission. Illegal transmissions are especially harmful on the VHF distress channel 16. Of course, unauthorized transmissions are performed anonymously. Reliable automatic identification of such transmissions could avoid the violation of radiotelephone regulation.

Another advantage of automatic identification follows from the ability of digital information inputting to another ships' navigational and information systems, for example Electronic Chart Display and Information System (ECDIS). ECDIS makes visualization of neighboring vessels in the range of VHF radio (i. e. approximately 30 nautical miles). However the transmitting vessel by no means is marked in an electronic map. Automatic identification would implement the vessel marking at electronic chart display.

One more application of AW is a covered information transmission in the special applications (for example, facing the threat of terrorist aggression).

It is essential that AW based identification doesn't require altering an existing radio installation and operational procedures. To introduce AW-identification function only new telephone receiver with the embedded processor at the transmitter side and processor with mini-display switched to common audio output at the receiver side have to be mounted. Automatic identification starts right away PTT pressing and runs continuously during all transmitting period independently from voice signal occurrence. No additional time and frequency channel recourses are required.

AW-identification provides the full compatibility with the existing transceivers and makes possibility of step-wise implementation.

Similar application of AW may be implemented in the aeronautical (118…136) MHz mobile service. In paper [3] watermarking based on speech unvoiced phonemes recognizing and replacing them by certain noise like sequence is proposed.

Maritime AW based identification is proposed in [4], [5]. In these papers watermarking process doesn't require any recognizing algorithms and based only on statistical signal properties.

In this paper we present stealthy AW forming algorithm based on signal energy saving and complete algorithmic ARTIS scheme for maritime application.

2 INFORMATION EMBEDDING ALGORITHM

2.1 *Audio watermarking system and its characteristics*

AW for maritime analog radiotelephony has some important features. First, watermarks are inaudible. Second, watermark technology doesn't demand any additional time and frequency resources beyond that used in the basic telephone channel. Thirdly, watermarks besides identification may convey also another digital. Finally, watermarking doesn't demand alteration of standard radio installation and operational procedures. Moreover, looking forward towards digital communications, voice watermarking straight away in the microphone could already solve identification problem.

Watermarking system is characterized by some competitive properties: fidelity, data payload and robustness [6]. *Fidelity* defines an audible similarity between an original voice message and watermarked message after information embedding. *Data payload* refers to number of bits a watermark encodes within a unit of time. *Robustness* is the ability of watermarks to survive the voice signal processing operations.

2.2 *Model of watermarked communication channel*

There are two approaches for information embedding in audio signal: a) spread spectrum (SS) method and b) quantization method. The first method is based on independent from the audio (host) signal forming of watermarking signal. Embedded information bit is spread by multiplying on certain pseudo random sequence (PRS) like in mobile communication. Herewith duration of elementary chip comes to inverse value of sampling frequency. On the contrary quantization method relays on the host signal.

Model of communication channel with additive watermarking is presented in Fig. 2. The encoder may be realized in the form of a) blind (or noninformed) or b) informed encoder depending on ignoring or using the information about the host signal x. The variant b) is reflected with dashed line. The watermarked signal s is additively formed from signals x and w:

$$s = x + w \tag{1}$$

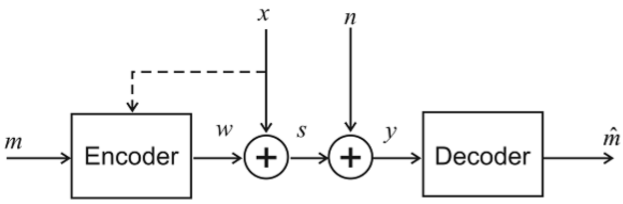

Figure 2. Model of watermarked communication channel

Power σ_w^2 of watermarked signal w is limited by the acceptable level of introduced distortions of carrier signal because watermark presence should be not audible (or quite tolerable) on the background of carrier signal x.

In the channel according the Fig. 2 two interferences act against watermark w: the first interference is itself the carrier signal with the power σ_x^2, and the second component – a noise n with the power σ_n^2.

Watermarking channel is characterized by its capacity – the maximum achievable code rate. Assuming the both interferences are white Gaussian noises the capacity C (bit/sample) of watermarked channel with noninformed encoder, when host signal is not available to encoder, is defined by Shannon's formula:

$$C = \frac{1}{2}\left(1 + \frac{\sigma_w^2}{\sigma_x^2 + \sigma_n^2}\right). \tag{2}$$

Practically $\sigma_x^2 \gg \sigma_n^2$, and capacity is limited mainly by the host itself.

At the same time using information about the carrier signal x, it is possible to increase C for informed encoder. In the paper [7] it was shown that assuming the host is known at the transmitter, the capacity of watermarked channel is defined by the formula:

$$C = \frac{1}{2}\left(1 + \frac{\sigma_w^2}{\sigma_n^2}\right). \tag{3}$$

Formula (3) shows that appropriately considered carrier signal doesn't influence on watermark transmission and the capacity is determined only by the second noise, which is unknown at the encoder. Capacity for such channel is increasing greatly, that's why an informed encoding (i.e. "writing on dirty paper") is attractive method for watermarking on account of its potential capacity.

2.3 *One channel encoding algorithm*

Let $\boldsymbol{x} = (x_1, x_2, \ldots, x_L)$ and $\boldsymbol{u} = (u_1, u_2, \ldots, u_L)$ are a host signal vector and binary PRS: $u_i = (-1, 1)$ in Euclidian L-dimensional space with norm and scalar product defined as follows

$$\|\boldsymbol{x}\| = \sqrt{x_1^2 + x_2^2 + \ldots + x_L^2}, \tag{4}$$

$$(\boldsymbol{x}, \boldsymbol{u}) = x_1 u_1 + x_2 u_2 + \ldots + x_L u_L. \tag{5}$$

It should be noted that Euclidian norm physically presents square root from the signal energy.

The task of encoder and decoder in the general scheme in Fig. 2 may be expressed in the next manner. Encoder must embed one bit of information $m = (-1, 1)$, i.e. it must watermark the vector $\boldsymbol{x}$, forming a certain new vector $\boldsymbol{s} = (s_1, s_2, \ldots, s_L)$ under provision the following conditions:

$$\|\boldsymbol{x} - \boldsymbol{s}\| = \min, \tag{6}$$

$$\|\boldsymbol{x}\| = \|\boldsymbol{s}\|, \tag{7}$$

$$(\boldsymbol{s}, \boldsymbol{u}) = Q(\tilde{x}, m, \Delta), \tag{8}$$

where $\tilde{x} = (\boldsymbol{x}, \boldsymbol{u})$, $Q(\cdot)$ - quantization function.

The target function (6) is grounded by minimization of introduced distortions because of watermarking process. Actually, under the equal conditions the less distortion, the stealthier is the watermarking signal $\boldsymbol{w} = \boldsymbol{x} - \boldsymbol{s}$.

Limitation (7) results from containing the energy of watermarked signal. Keeping the signal energy gives an ability to apply quantization function with adaptive step proportional to the power of the host signal in the encoder and received signal in the decoder. Adaptive quantization eliminates influence of amplitude scaling on watermarks. Note that such adaptive quantization is equivalent to normalization by signal norm and quantization with the constant step. Watermarking scheme based on quantization of normalized correlation between the host vector and a random vector is proposed in the paper [8] for images watermarking problem.

Optimization problem (6) subject to limitations (7), (8) in general needs rather difficult analytical solution and sizeable computations. The desired optimal solution may be easily obtained according to the next argumentations.

Having the host signal $\boldsymbol{x}$ it is possible to compute $\tilde{x}$ and then its quantized version $\tilde{s} = Q(\tilde{x}, m, \Delta)$ subjected to information bit m and quantization step Δ. Quantization function $Q(\cdot)$ has adaptive step proportional to the host signal norm: $\Delta = k\|\boldsymbol{x}\|$, where $k = (0.5 \ldots 2)$ assigns distortions level. To be specific $m = (-1,\ 1)$ agrees with odd and even quantization levels correspondingly. Quantization step adaptation provides immunity from amplitude scaling and contributes to watermarks audio imperceptibility in the speech silent intervals.

From the well-known relation $(\boldsymbol{s},\boldsymbol{u}) = \|\boldsymbol{s}\|\|\boldsymbol{u}\|\cos(\vartheta)$ angle ϑ between vectors $\boldsymbol{s}$ and $\boldsymbol{u}$ (see Fig. 3) may be obtained as:

$$\vartheta = \cos^{-1}(\tilde{s}/\|\boldsymbol{x}\|\|\boldsymbol{u}\|) . \tag{9}$$

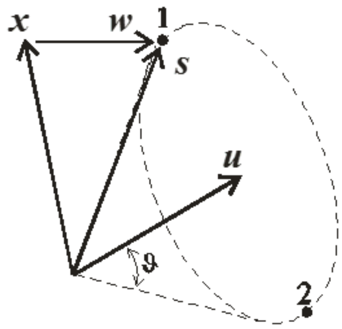

Figure 3. Geometric interpretation of norm saving data embedding in three dimensional space

The last formula takes into account relations $\|\boldsymbol{x}\| = \|\boldsymbol{s}\|$ and $\tilde{s} = (\boldsymbol{s},\boldsymbol{u})$. Required vector $\boldsymbol{s}$ lies on a surface of L-dimensional cone that has center axis $\boldsymbol{u}$ and subtending angle 2ϑ. The cone is shown by dashed lines.

From the other hand to minimize the embedding distortion we need find the closest vector on this cone to the host vector $\boldsymbol{x}$. It is clear that vector $\boldsymbol{s}$ lies in the plane which is formed by vectors $\boldsymbol{x}$ and $\boldsymbol{u}$. Therefore vector $\boldsymbol{s}$ may be expressed by the combination of $\boldsymbol{x}$ and $\boldsymbol{u}$, namely

$$\boldsymbol{s} = \alpha\boldsymbol{x} + \beta\boldsymbol{u}, \tag{10}$$

where α and β are embedding factors to be found.

To do this let us multiply scalar wise by $\boldsymbol{u} \neq 0$ the two parts of relation (10). After evident transformations we get

$$\beta = (\tilde{s} - \alpha\tilde{x}) / \|\boldsymbol{u}\|^2 . \tag{11}$$

Then again multiplying relation (10) scalar wise by $\boldsymbol{s} \neq 0$, and solving quadratic equation, we get

$$\alpha_{1,2} = \pm\sqrt{\frac{\|\boldsymbol{x}\|^2\|\boldsymbol{u}\|^2 - \tilde{s}^2}{\|\boldsymbol{x}\|^2\|\boldsymbol{u}\|^2 - \tilde{x}^2}} . \tag{12}$$

The nearest to $\boldsymbol{x}$ vector $\boldsymbol{s}$ must be taken with the positive sign for α in (12). This vector corresponds to point 1 in Fig. 3. On the contrary negative sign in formula (12) gives the most distant vector $\boldsymbol{s}$ (point 2).

Generally embedding may be realized in the time or frequency domain after Fast Fourier Transformation (FFT). The last choice is more preferable because of less watermark sensitivity to time shifts within sampling interval. In the frequency domain the amplitudes of FFT coefficients must be used for information embedding.

Mathematically amplitudes are expressed by nonnegative numbers. But if undertake nothing above presented algorithm might give negative coordinates in $\boldsymbol{s}$ especially for low power vector $\boldsymbol{x}$. To prevent this we should add to previously introduced limitations (7), (8) one more inequality limitation

$$s_i \geq 0, i = 1, 2, \ldots, n . \tag{13}$$

For this additional limitation suboptimal algorithm for minimizing $\|\boldsymbol{x} - \boldsymbol{s}\|$ is derived. Appropriate pseudo code is given below

while min(s_i) < 0, i=1 ... n

x = abs(s);

Get new $\tilde{x}, \alpha, \beta, \boldsymbol{s}$; (according (10) – (12))

end

Practically this algorithm already converges just on the second step.

Informed encoding eliminates interference on watermarking signal from the host signal but needs a certain time delay at the transmitter side.

2.4 *Decoding algorithm*

Received vector is given in the form:

$$\boldsymbol{y} = \boldsymbol{x} + \boldsymbol{w} + \boldsymbol{n} \tag{14}$$

where $\boldsymbol{n}$ – additive noise vector.

Decoder computes scalar product $\tilde{y} = (\boldsymbol{y},\boldsymbol{u})$ and then extracts the embedded bit by applying

$$\hat{m} = \arg\min_{m=(-1,1)} \left|\tilde{y} - Q(\tilde{y}, m)\right| . \tag{15}$$

For binary quantization decision (15) may be reduced to $\hat{m} = sign(\tilde{y})$.

3 INTERFERENCES AGAINST WM

Besides host signal interference there are another interferences which are essential for watermarks restoration at the receiver. These interferences in the watermarking theory are identified as attacks. The most harmful attacks are:

1 intersymbol interference (ISI);
2 amplitude scaling;
3 additive noise;
4 nonlinear distortions (clipping);
5 resampling and desynchronization.

Hofbauer et al. [9] proposed to take into account Doppler effect that is actual for aeronautical applications, but in the maritime communication it might be neglected.

3.1 *Intersymbol interference*

In the radio channels ISI is usually caused by multipath propagation. The transmitting medium in VHF radio communication is the atmosphere, in which radio signal is transferred by means of electromagnetic waves. The received electromagnetic signal is usually a superposition of a line-of-sight path signal and multiple waves coming from different directions. This phenomenon is known as multipath propagation. The received signal is spread in time and the channel is said to be time dispersive.

Another physical cause of ISI is nonuniformity of frequency response of a channel. Analog low-frequency circuits of the transceiver are composed from reactive elements. Frequency dependent elements cause nonuniformity of frequency response within audio signal spectrum. When frequency response is explicitly nonuniform within signal spectrum output signal is highly differs from input one. Distortions caused by bandlimited low-frequency channel also represent ISI.

From the signal processing point of view the two physically different causes (presence of reactive elements in audio circuits and multipath radio wave propagation) lead to the same final result in the form of ISI.

The most effective measure against ISI is orthogonal frequency division multiplexing (OFDM) technology that is commonly used in numerous communication systems.

3.2 *Amplitude scaling*

Amplitude scaling refers to uncertainty of incoming signal amplitude and its slow variations. Coming back to multipath propagation, one can analyze a variant when the different path lengths are very similar compared to the wavelengths of the signal components. Then the phase variations between components will be small and they will all undergo very similar amounts of cancellation or reinforcement. This case is usually termed flat fading. In watermarking flat fading is simulated by amplitude scaling attacks.

In general quantization based methods are very sensitive to such attacks. Counteractions against amplitude scaling are adaptive step quantization or double signal normalization before and after quantization to the signal norm. Application of these operations demand saving the signal norm in the watermarking process.

The proposed algorithm is exactly designed for saving the signal norm (or squared root signal energy).

3.3 *Additive noise*

Additive noise is imposed onto the signal during transmission. The noise results from thermal noise in electronic circuits, from atmospheric noise or from other radio stations. Quantization noise from analog-to-digital converter may be attributed to additive noise. Commonly recognized model of an additive noise is additive white Gaussian noise, denoted in Fig. 2 by n.

Additive noise is measured by signal-to-noise ratio (SNR) which practically comes to (15…17) dB.

3.4 *Nonlinear distortions (clipping)*

Nonlinear distortions appear in amplitude limitations caused, for example, by the overload in audio circuits. Overload arises from redundant power of transmitting station. The simplest model of nonlinear distortions is clipping. Radiotelephone AW in any case should be resistant against such distortions.

3.5 *Resampling and desynchronization*

At the transmitter and receiver sampling processes are not synchronized. It means that sampling instants in the receiver are shifted relatively corresponding samples in the transmitter. Consider sampling frequencies are equal at the transmitter and receiver. So resampling comes to arbitrary shift of sampling points. Desynchronization is here understood as uncertainty in the starting of watermark in the receiver. For correct watermark separation the beginning of watermark should be first detected and then all decision points are counted from the starting point. Analog radiotelephone channel by all means leads to resampling and loss of the watermark beginning.

Watermark sensitivity to resampling considerably depends on choosing of embedding domain – time or frequency. Assume signal samples are independent. Correlation time in this case may be accepted to be $\tau_c = 1/(2F_s)$, where F_s is sampling frequency. Frequency domain embedding is based on FFT computation which uses, for instance, N consecutive samples. FFT amplitudes in every frequency channel have in this case N time enlarged correlation period: $\tau_c = N/(2F_s)$. Therefore frequency domain WM is considerably less sensitive to sampling shifts than time domain WM.

4 PRACTICAL REALIZATION

4.1 *OFDM based AW embedding*

AW are multiplexed directly into audio signal and therefore subjected to influence of common for audio signal transformations and interferences. Standard transformations are: amplification, modulation, filtering. In the standard band limited audio channel 300 – 3000 Hz ISI impacts destructively on watermarks.

ISI is caused mainly by nonuniformity of channel frequency response and nonlinearity of phase response in low frequency transceiver electronic circuits. The most effective measure against ISI is orthogonal frequency division multiplexing (OFDM) technology that is commonly used in numerous communication systems.

OFDM approach for AW was proposed in [10] and. The main idea lies in embedding every watermarking bit completely into a certain narrowband component of host signal. AW information jointly with host signal form OFDM symbol.

Construction of OFDM symbol for AW is illustrated in Fig. 4. For simplification the figure is drawn for the next numerical values: FFT dimension $N=8$; length of spreading sequence $L=5$; number of watermarked subchannels $B=2$.

Vector $\boldsymbol{x}$ of host audio samples in the time domain, a) is buffered by columns into $N \times L$ matrix, b); FFT along each column is performed, c); B watermarking bits are embedded in the frequency coefficients independently along rows, d). In Fig. 4 samples in the time and frequency domains are shown by squares and circles accordingly; virgin samples are shown white and modified ones due to watermarking are grey. As it seen watermark distortions are distributed along the whole watermarked sequence $\boldsymbol{s}$.

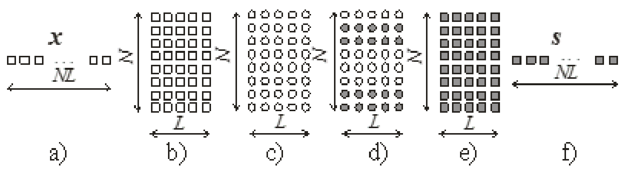

Figure 4. Forming of OFDM watermarking symbol

Embedding algorithm for each subchannel are presented in section 2.

General system for informed OFDM encoding/decoding is presented in Fig. 5. Signal $\boldsymbol{x}$ is split in serial-to-parallel (S/P) demultiplexer into N slow flows which are subjected to FFT. Message flow m is split into some, suppose B, $B < N/2$ slow flows. Channel encoders use B Fourier coefficients for watermarking independently in each channel. In the simple case one bit may watermark one coefficient. For more complex variants one embedded bit may be distributed among $L>1$ coefficients. In general NL samples of $\boldsymbol{x}$ are needed for embedding B message bits. Algorithms for each channel encoders are identical. Watermarked coefficients and all the rest undisturbed coefficients are then used for inverse FFT (IFFT). Again parallel-to-serial (P/S) block combines slow flows into one sequence $\boldsymbol{s}$ in the time domain.

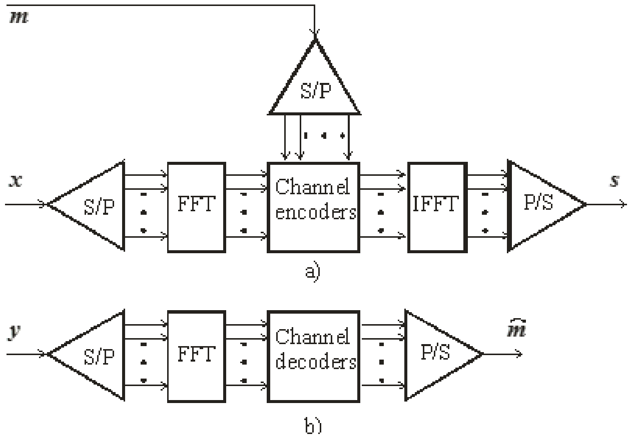

Figure 5. Multichannel realization: a) encoder, b) decoder

At the receiver signal $\boldsymbol{y}$ is split into N flows which are transformed in Fourier coefficients. These coefficients are processed according demodulation algorithm for extracting a watermark message bits $\hat{m}$.

Parameters N, L, B, Δ are subject for optimal tradeoff decision fidelity – data payload – robustness. Data payload is expressed in bit/sec by the formula

$$R = BF_s/(NL). \tag{16}$$

Parameter L influences on watermarking fidelity. The greater L, the higher fidelity is. Objectively fidelity may be expressed by Watermark-to-Signal Ratio

$$WSR = 20lg\left(\sigma_w/\sigma_x\right). \tag{17}$$

Embedding and detection processing for simultaneous multichannel processing according to OFDM principle is well realized in the matrix form that doesn't need using any long duration cycles.

Watermarking signal $\boldsymbol{w}$ in OFDM presentation occupies the whole frequency audio band and simulates an ordinary additive noise. By appropriate choosing of parameters in the embedding algorithm it is possible to eliminate audibility of $\boldsymbol{w}$ on the background of host signal and external noise.

It should be noted very good stealthy characteristics for truncated vector $\boldsymbol{x}$ of length $L=2$. For this case embedding distortions, expressed by WSR, may be much more numerically increased comparatively with large L. It is explained by concentration of one bit watermark signal in the short time interval (about 16 msec under conditions $F_s = 8$ kHz, $N=64$). Within this interval the total audio energy is preserved due to embedding algorithm and energy differences between host signal $\boldsymbol{x}$ and watermarked signal $\boldsymbol{s}$ inside the short speech interval are not perceived. Therefore watermark energy may be gained under the same audible similarity. At the same time L reduction comes to increasing data payload according to formula (16) and simplifies technical realization.

Variant of ARTIS constructer design is presented in Fig. 6.

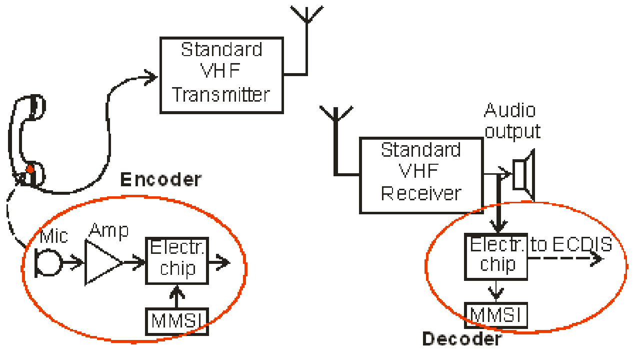

Figure 6. Example of VHF ARTIS design

4.2 *AW detection and synchronization*

Estimation of watermarking bits (15) needs proper synchronization the correlation process. Synchronization and marker fields consume a certain time and decrease an effective information bitrate. Additional resources are spent also for check sum.

To settle all these tasks simultaneously we propose to compute sliding correlation and detect watermark by detecting the whole OFDM symbol using hash function. In general hash function transforms an input data of variable size to a fixed-size string, which is called hash value. Commonly hash function is employed for checking integrity of input data. We use hash function for the purpose of OFDM watermarking symbol detection. Integrity of the detected symbol is ensured automatically.

To implement this idea let us compose watermarking data containing information bits M itself and its hash function $H(M)$: $[M, H(M)]$. On the receiving part decoder is permanently processing blocks of samples until hash function will satisfy.

Suppose current block $[s_i,\dots,s_{i+NL-1}]$, containing NL samples, gives estimations of information bits $\hat{M}$ and received hash function $\hat{H}(M)$. Decoder calculates hash function over the received information bits $H(\hat{M})$. If $H(\hat{M})=\hat{H}(M)$ then the decision of watermark symbol detection is accepted and the detected bits are believed $\hat{M}$. Otherwise next incoming block $[s_{i+1},\dots,s_{i+NL}]$ enters for processing.

Let us estimate a sufficient length of hash-function. "Good" hash function maps every input combination into unique output combination. Then lengths sequences of information bits M and hash function $H(M)$ should coincide. Suppose $length(M)=length(H)=l$. False detection will take place if and only if some random l-length input combination will give l-length hash function that coincides with the subsequent bits. The probability of such event is $p_{er}=2^{-2l}$. Taking $l=8$ one can obtain $p_{er}=2^{-16}\approx 1.5*10^{-5}$ for randomly occurred sequence. Of course, total false detection will increase proportionally the searching time t_s for subsequent AW symbol. To reduce t_s encoder is blocked after successive detection at the time slightly below the symbol duration (NL samples in our notation).

Practically cyclic redundancy check (CRC) code of length $l=8$ CRC-8 with generator polynomial x^8+x^2+x+1 may be used.

The proposed detection method in our application exceeds standard communication format of fields: synchronization – marker – data – CRC in the useful watermarked information per host sample. This method provides decoding OFDM watermarking symbol in the whole and thus saves from the necessity of synchronization and marker fields at all. Second advantage is the absence of synchronization error on correlation receiver accuracy.

These superiorities are achieved however at the expense of considerable processing loading in the receiver. But detection algorithm is based mainly on FFT matrix operations and may be easily on-line realized for audio frequency.

5 EXPERIMENTAL AND SIMULATION RESULTS

Computer simulation in MatLab environment was held using for host signal the real voice frame of length 27500 samples. Parameters of simulation are presented in Table 1.

Table 1. Parameters of simulation and experiment

Parameter, notation	Numerical value
Sampling frequency, F_s	8.0 kHz
OFDM sub-channel width	125 Hz
Number of sub-channels, B	15
WSR	- 16/-14/-12 dB
SNR	12 dB
AW raw rate, R	60/125/268 bit/sec
m-sequence length, L	31/15/7
FFT dimension, N	64
Delay in the transmitter, D_{Tx}	56/120/248 msec
Identification time	0.25/0.5/1 sec

In the Table 1 the following notations are taken:

$WSR=20lg(\sigma_w/\sigma_x)$ - Watermark-to-Signal Ratio, dB;

$SNR=20lg(\sigma_s/\sigma_n)$ - Signal-to-Noise Ratio, dB;

$D_{Tx}=NL/F_s$ - Delay time in the transmitter, sec;

$R=BF_s/NL$ - AW raw data payload, bit/sec.

Identification time is estimated taking into consideration the total number of bits, needed for representation of 9-symbol decimal MMSI (Maritime Mobile Service Identity). This value is accepted by 30 bits. CRC length is taken $l=8$.

Simulation have being carried taking into consideration all interferences and limitations, discussed in Section 2.

Experimental testing was carried out in the VHF channel 17 of maritime mobile service (156.85 MHz); emission class F3E/G3E (frequency/phase modulation, analog telephony) on the base of hardware installation: maritime VHF transceivers RT-2048 Sailor, USB ADC/DAC (analog-to-digital converter/digital-to-analog converter) module E14-140 L-CARD.

In the trials all watermarking symbols along the voice frame were detected correctly and no false detected symbols were registered.

6 CONCLUSION AND FUTURE INVESTIGATIONS

The addressed and properly understood radiotelephone communication in VHF, MF and HF bands plays an important role in general maritime safety. Significance of particularly VHF channel 16 was emphasized, for instance, in the paper [11]. Automatic identification, in turn, would provide an effective and clearly understood communication. The considered automatic identification solves a number problem towards VHF radiotelephone improvement, elimination a human factor and finally enhancement of maritime safety and security. As opposed to river system [1], ARTIS is grounded not on the appending a certain digital sequence to the radiotelephone transmission, but realizes just its identification from the very beginning of the transmission, and permanently runs during the whole transmission.

AW identification doesn't require an additional frequency and time resources, alteration standard transceivers and radio communication procedures and appears only in additional noise, that can be set to minimal level (equal or below of the channel noise).

Implementation the proposed ARTIS function needs only phone receiver replacement by a new one with built-in processor at the transmitter side and switching a decoder to standard audio output at the receiver part. It easily provides compatibility of standard equipment and the equipment with identification function. Moreover, AW identification works as well in digital channels if the last will start operating in maritime radiotelephony. This ability is ensured by the AW resistance against ADC/DAC and voice compression procedures.

ARTIS implementation makes possible a further integration of VHF radio and navigation (AIS/ECDIS) equipment.

The next efforts in ARTIS development should be steps in hardware realization for real time processing and investigations of ARTIS adaptability in the MF/HF bands of maritime communication.

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