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到来方向情報に基づくデジタルビームフォーマと変調方式自動認識/復調モジュールを搭載した電波監視システム

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あらまし 到来方向情報に基づくデジタルビームフォーマと変調方式自動認識/復調モジュールを搭載した電波監視システムを提案する．本システムは現行電波監視システムと同様にはじめにアレーアンテナを用いて電波の到来方向を同定する．次にこの到来方向情報を用いて特定の信号のみを抽出するビームフォーミングを施す．この前処理により混信状況を回避できるため従来より提案されている変調方式自動認識アルゴリズムを確度高く動作させることができる．最後に推定された変調方式に相当する復調ソフトウェアにビームフォーミング出力を入力することによりターゲット信号と到来方向情報のペアを監視することができる．さらに本報告では本システムを DSP に実装しリアルタイム処理の可能性を検討した．

キーワード 電波監視システム, 変調方式自動認識/復調, 到来方向推定, ゼロフォーシングビームフォーマ

DOA based Signal Combining aided Automatic Modulation Recognition/Demodulation for Surveillance System

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Abstract This paper presents automatic recognition technique of analog modulation scheme and signal waveform detection of the signals in a surveillance system. This system consists of array antenna signal processor and automatic signal recognition/demodulation modules. Firstly, directions of arrival (DOA) of the incoming signals are estimated by using super-resolution algorithm. Using these DOA estimates, the respective signal is beamformed while nulling out the others and thus increase SINR of the system. Higher SINR will therefore improve recognition detection performance of the system. Automatic modulation recognition, which is based on decision-theoretic approach, followed by demodulation of the signals, is carried out on the extracted signals to analyze the characteristics of the signals.

Key words surveillance system, automatic modulation recognition/demodulation, directions of arrival estimation, zero-forcing beamformer

1. Introduction

Wireless communication has become a hot topic of research among researchers over these few years and it is predicted that wireless will overtake wired in next few years as main communication system in our daily life.

Nevertheless, such activities involve the use of limited resource, that is, radio waves. Radio waves, if the rules are not abided by, could cause severe intermodulation and interference. According to the data[1] released by Ministry of Public Management, Home Affairs, Posts and Telecommunications, Japan, number of unlicensed radio stations activities has increased and this trend will continue with increased number of wireless appliances. Current system that is used to monitor unlicensed signal, however, only detect the location of the signal without analyzing the content of the signal itself.

Our proposal, in an aim to replace current DEURAS-H system†, will able to simultaneously detect the directions of arrival (DOA), modulation type as well as the content of the transmitted signal in real time. The system, which comprises 7 sensors in uniform circular array antenna (UCA), firstly estimates the DOA of the signals by using UCA-ESPRIT[2][3] algorithm, then based on the knowledge of DOA information the target signal is extracted by using zero-forcing beamforming algorithm and passed through an automatic modulation recognition module[4][5]. Finally, the output signal of the beamformer is demodulated according to the result of automatic modulation recognition algorithm. Compared with [6], that implements the DOA estimator and automatic modulation recognition/demodulation modules independently, our proposal systematically integrate the modules such as, DOA estimator, signal combiner, and automatic modulation recognition/demodulation modules, that results in the performance improvement of the system.

2. System Architecture

Our proposed surveillance system comprises five main modules, that is, Array Receiver, DOA Estimator, Signal Combiner, Automatic Modulation Recognizer and finally Demodulator modules as in Fig.1. Here, we have assumed that carrier frequency analyzer and frequency selection filter are implemented in the array receiver. In this paper, we would like to focus on the four modules after the array receiver.

2.1 DOA Estimation

A planar array is needed if estimates of source azimuth

†Direction finder that operates at low frequency range, which usually ranges from 300 kHz to 30 MHz (HF signals), where the signals are mainly analog modulated signals such as AM, DSB, SSB and FM signals.

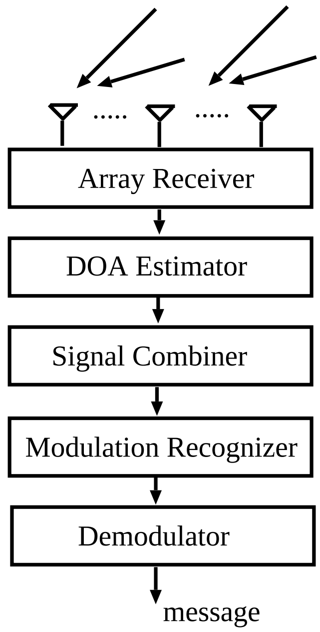

Fig. 1 Flowchart of proposed surveillance system.

and elevation angles are required. The following property of uniform circular arrays (UCAs) makes them attractive in the context of DOA estimation. UCAs provide 360° azimuthal coverage, and also provide information on source elevation angles.

UCA-ESPRIT, a technique used to estimate the DOA of incoming signal, is employed in our proposed system. UCA-ESPRIT does not posses the displacement invariance required by ESPRIT-based algorithms[7]. The similarity to ESPRIT is in that the eigenvalues of a matrix directly yield the DOA estimates. The eigenvalues have the form $\mu_i = \sin\theta_i e^{j\phi_i}$, where θ_i and ϕ_i are the elevation and azimuth angles of the ith source, respectively.

2.2 Signal Combining

Here, we assume that single desired signal denoted by subscript '1' and single interfering signal denoted by subscript '2' as in Fig.2.

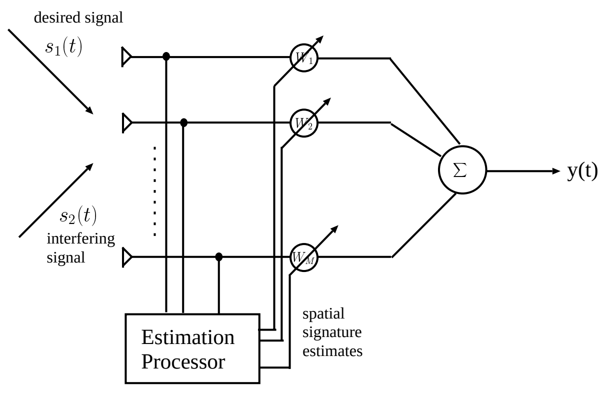

Fig. 2 Optimal signal combiner.

Given the weight vector $\boldsymbol{\omega}$, the signal estimate at the array output can be obtained by

$$y(t) = \boldsymbol{\omega}^H \boldsymbol{x}(t) = y_1(t) + y_2(t) + y_n(t) \quad (1)$$

where $\boldsymbol{x}(t)$ is the array input vector and $y_1(t), y_2(t), y_n(t)$ are the contributions from the desired signal, the interfering signal, and the noise, respectively. They are given by

$$y_1(t) = \boldsymbol{\omega}^H \boldsymbol{v}(\theta_1, \phi_1) s_1(t) \quad (2)$$
$$y_2(t) = \boldsymbol{\omega}^H \boldsymbol{v}(\theta_2, \phi_2) s_2(t) \quad (3)$$
$$y_n(t) = \boldsymbol{\omega}^H \boldsymbol{n}(t). \quad (4)$$

where $\boldsymbol{v}(\theta, \phi)$ is an array mode vector corresponds to the signal impinging from the elevation angle θ and the azimuth angle ϕ.

If the signals are modeled as unknown and deterministic, and the noise is modeled as temporally and spatially white complex Gaussian, then the maximum likelihood estimates of the signals are given by

$$\hat{s}(t) = \arg\min_{s(t)} ||\boldsymbol{x}(t) - \boldsymbol{V}\boldsymbol{s}(t)||^2 \quad (5)$$

where $\boldsymbol{V} = [\boldsymbol{v}(\theta_1, \phi_1), \boldsymbol{v}(\theta_2, \phi_2)]$ and $\boldsymbol{s} = [s_1(t), s_2(t)]^T$. Equation (5) is identical to the least square estimates. In the case that $s(t)$ is a continuous function and $\boldsymbol{V}$ is the column full rank matrix, the solution is

$$\hat{s}(t) = \boldsymbol{V}^{(+)} \boldsymbol{x}(t) \quad (6)$$

where $\boldsymbol{V}^{(+)} = (\boldsymbol{V}^{(H)}\boldsymbol{V})^{-1}\boldsymbol{V}^H$ and $\boldsymbol{V}^{(+)}\boldsymbol{V} = \boldsymbol{I}$.

The resulting optimal weight vector to estimate the desired signal can be obtained by

$$\boldsymbol{\omega}_{opt} = \boldsymbol{V}(\boldsymbol{V}^H\boldsymbol{V})^{-1}\boldsymbol{u}_1 \quad (7)$$

where $\boldsymbol{u}_1 = [1, 0, \cdots, 0]^T$ denotes the 1st column of identity matrix.

2.3 Automatic Modulation Recognition

A functional flowchart to discriminate and classify different modulation types of interest can be illustrated in Fig.3. As shown in the figure, three key features have been considered in our system and a brief explanation of these features will be given here.

2.3.1 $\gamma_{\max}$

The first key feature, γ_{max} is defined by

$$\gamma_{\max} = \frac{\max|\mathrm{DFT}(a_{cn}(i))|^2}{N_s} \quad (8)$$

where N_s is the number of samples per segment and $a_{cn}(i)$ is the value of the normalized-centered instantaneous amplitude at time instants $t = \frac{i}{f_s}$, (i=1,2,. . .,N_s), and it is defined by

$$a_{cn}(i) = a_n(i) - 1 \quad (9)$$

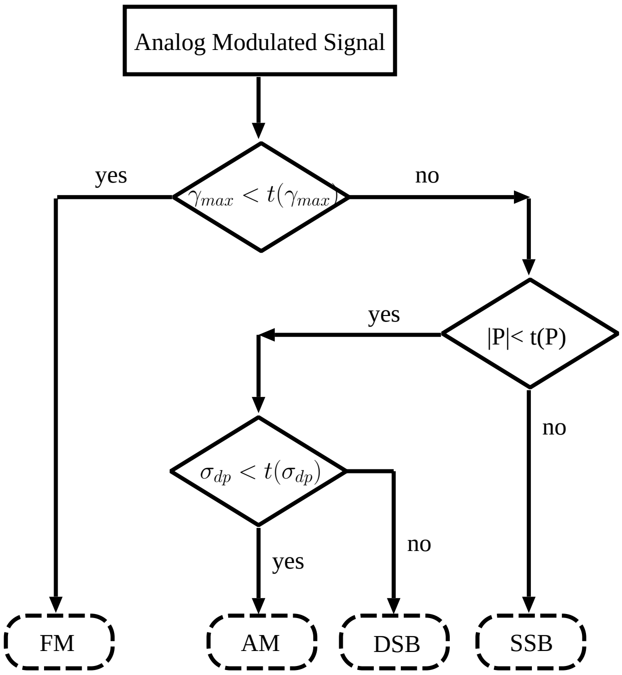

Fig. 3 Funtional flowchart for modulation recognition.

where $a_n(i) = |y(i)|/m_a$ and m_a is the average value of instantaneous amplitude evaluated over one segment

$$m_a = \frac{1}{N_s}\sum_{i=1}^{N_s} |y(i)|. \quad (10)$$

This value is used to discriminate between FM signals as first subset and AM, DSB, SSB as the second subset. As the FM signals are considered to have constant instantaneous amplitude in a segment, their normalized-centered instantaneous amplitude is zero. Thus, their spectral power densities are also zero. It could be confirmed from Fig.4. It should also be noted that the distance of $\gamma_{\max}$ between the first and second subsets depends on the SNR or almost equivalently SINR, i.e. higher SINR achieves larger distance of $\gamma_{\max}$.

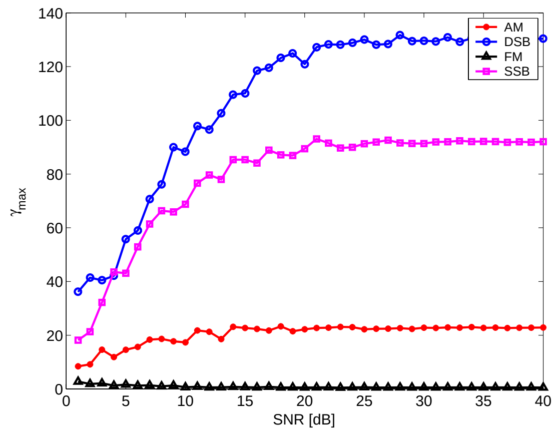

Fig. 4 Dependence of $\gamma_{\max}$ on SNR.

2.3.2 P

The second key feature is used for measuring the spectrum symmetry around the carrier frequency, and it is based on the spectral powers for the lower and upper sidebands of the signal. Mathematically, it is defined as

$$\mathrm{P} = \frac{P_L - P_U}{P_L + P_U} \tag{11}$$

where,

$$P_L = \sum_{i=1}^{f_{cn}} |Y_c(i)|^2 \tag{12}$$

$$P_U = \sum_{i=1}^{f_{cn}} |Y_c(i + f_{cn} + 1)|^2 \tag{13}$$

$Y_c(i)$ is the Fourier transform of the signal y, $(f_{cn} + 1)$ is the sample number corresponding to the carrier frequency, f_c and f_{cn} is defined as

$$f_{cn} = \frac{f_c N_s}{f_s} - 1 \tag{14}$$

The ratio of P is used to discriminate between SSB as a first subset and AM, DSB as a second subset, since P at infinite SNR ought to be 1 for SSB signals (+1 for LSB and -1 for USB), and 0 for AM and DSB. It could be confirmed from Fig.5.

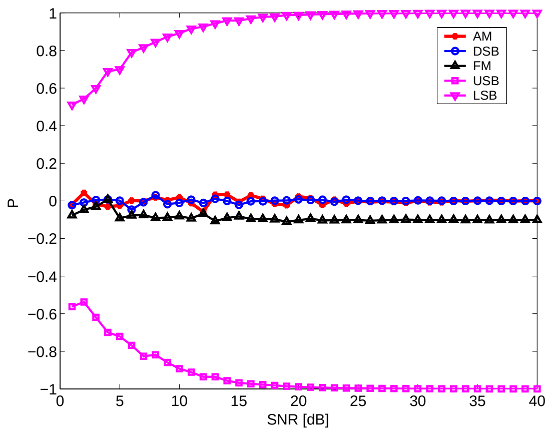

Fig. 5 Dependence of P on SNR.

2.3.3 σ_{dp}

The third key feature is defined as

$$\sigma_{dp} = \sqrt{\frac{1}{C}\Big(\sum_{a_n(i)>a_t} \phi_{NL}^2(i)\Big) - \Big(\frac{1}{C}\sum_{a_n(i)>a_t} \phi_{NL}(i)\Big)^2} \tag{15}$$

where $\phi_{NL}(i)$ is the value of the centered non-linear component of the instantaneous phase of signal $\arg\{y(t)\}$ at time instant $t = \frac{i}{f_s}$, C is the number of samples in $\phi_{NL}(i)$ for which $a_n(i) > a_t$ and a_t is a threshold for $\{a(i)\}$ below which the estimation of the instantaneous phase is sensitive to the noise. Thus, σ_{dp} is the standard deviation of the centered non-linear component of the direct instantaneous phase, evaluated over the non-weak intervals of a signal segment.

σ_{dp} is used to discriminate between AM and DSB. This is due to the fact that AM signal has no direct phase information. On the other hand, the other types of modulation have direct phase information. It could be confirmed from Fig.6.

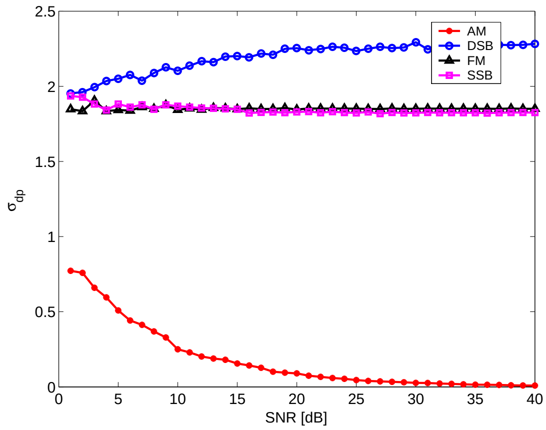

Fig. 6 Dependence of σ_{dp} on SNR.

2.4 Adaptive Demodulation

Analog demodulation is the process of taking the received signal and reconstructing an approximation of the transmitted message signal. There are several methods used for the demodulation of analog signals. In our approach, we only consider coherent demodulation, which performs well in noisy environment. Figure 7 illustrates adaptive demodulation process used in the proposed surveillance system. All of the demodulators are implemented in the software and chosen adaptively according to the decision of the automatic modulation recognizer.

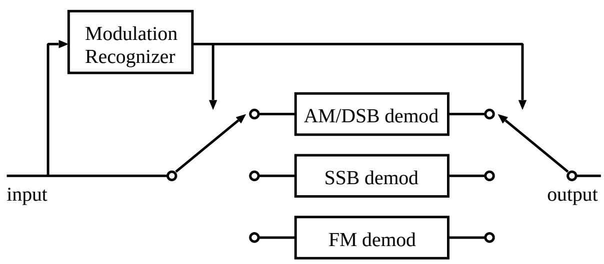

Fig. 7 Adaptive demodulation module.

3. Simulation Results

Table 1 shows general settings used in the simulation.

Besides that, threshold values for those key feature extractions is illustrated in Table 2.

Table 1 General settings.

Array Antenna	Circular
Sensors	7
Radius of the UCA	half wavelength
# of samples per segment	1024
IF frequency, f_c	18 [kHz]
Sampling rate, f_s	64 [kHz]
Noise	AWGN

Table 2 Threshold values setting for key feature extraction.

Key Features	Threshold Value, t(·)
γ_{max}	6
P	0.8
σ_{dp}	0.4

Let us assume that AM signal and FM signal with voice messages as shown in Fig.8, arrived from 50° and 130° in azimuthal angles 70° and 60° in elevation angles respectively in the same frequency band. SNR for both of the signals is set to be 15 [dB]. The voice message on top of Fig.8 represents modulating signal for AM while the bottom voice represents modulating signal for FM.

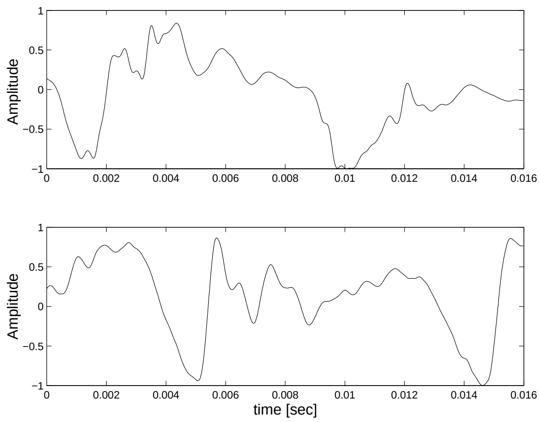

Fig. 8 Original voice signals for AM and FM.

In our proposed surveillance system, firstly the DOA estimation is performed and the results are shown in Fig.9. Assuming that the signal from azimuthal direction of 50° to be our signal of interest, secondly the zero-forcing beamforming is performed and the resulting array beam pattern is illustrated in Fig.10. It seems to be enough to increase the input SINR toward the modulation recognizer. As expected, the estimate of automatic modulation recognition algorithm is "FM". Finally the demodulated signal and the original signal are compared in Fig.11. Although the demodulated signal is still corrupted with noise even after the demodulation, we could conclude that the original signal could be reconstructed from the proposed surveillance system.

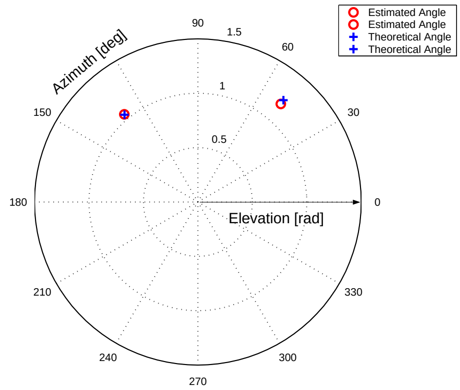

Fig. 9 Estimated azimuth and elevation angles.

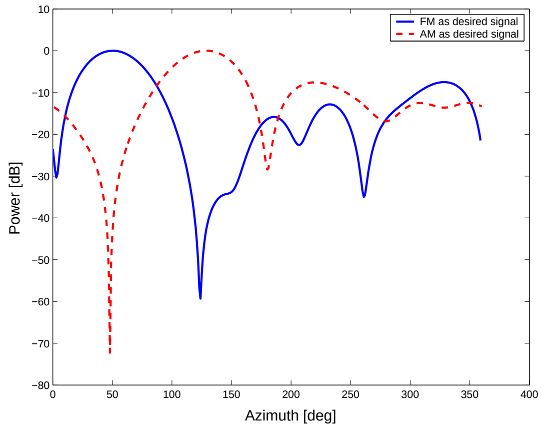

Fig. 10 Array beam pattern for extraction of the desired signal.

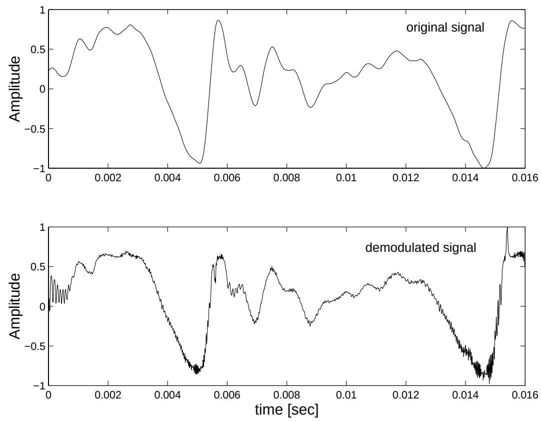

Fig. 11 Comparision of original and demodulated signals.

4. DSP Implementation

In this section, we will investigate the process time for the system both in CPU and DSP environment. In order to run the program in CPU and DSP environments, programs written in matlab has been rewritten into C program. This includes some calculation package such as complex eigenvalue decomposition and complex inverse matrix calculation from LAPACK[9].

Specification for CPU is expressed at Table 3. Meanwhile, Apex-Pro[8] has been used as the development and compilation tool in the implementation of DSP.

Table 3 Specification for CPU environment.

CPU	Pentium III 700 MHz
Memory	256 MHz
C Compiler	Visual C++ 6.0

Table 4 Estimated DOA in CPU and DSP environments.

	Theoretical	PC	DSP
Azimuth #1°	60.0	60.095474	60.078648
Elevation #1°	60.0	60.062662	60.062729
Azimuth #2°	30.0	30.011834	30.048889
Elevation #2°	30.0	30.010314	30.045544

Two incoming signals of 60° and 30° in azimuthal angle, 60° and 30° in elevation angle respectively are assumed to arrive at the array. The results in CPU and DSP are compared in Table 4. Results from CPU and DSP verified that DSP performed as good as in PC calculation.

The next step is to compare the process time both in CPU and DSP environments. Both environments used clock() function in C library to calculate the process time. In CPU, the process time is 90 [msec], while the process time for the same program in DSP is found out to be approximately 93 [msec]. The detailed process time in DSP is illustrated in Table 5. In this case the demodulation block seems to be a bottleneck. Although the implementation time for CPU is faster than in DSP, it doesn't mean that CPU outperforms DSP. To accomplish the realtime demodulation for the voice signal, we need smaller than 16 [msec] processing time for a segment in the demodulation block. Since the clock for now using SHARC DSP is 40 [MHz] and the demodulation block is implemented on the single DSP chip, we need 230 [MHz] DSP for realtime implementation, but 3.9 [GHz] in CPU. Considering this factor, we can conclude that DSP actually outperforms CPU in the real time implementation.

Table 5 Process time of each function in DSP.

	Process Time [msec]
DOA and Signal Combining	13
Automatic Modulation Recognition	18
Demodulation	62
Total	93

5. Concluding Remarks

Automatic modulation recognition/demodulation technique aided by DOA based signal combining was proposed for the next generation surveillance system. It does not only estimate the DOA of the target signals, but also detect the modulation type and extract the contents of the target signals. The proposed system was simulated under matlab environment and the results show that each module integrates well with each other in HF environment. Furthermore, the proposed system was implemented in DSP, and it showed us the possibility of the real-time processing.

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