BLISS: A Blind Spectrum Separation Approach for Jamming-Resistant Communications

Srikanth Pagadarai
Department of Electrical and Computer Engineering
Worcester Polytechnic Institute, Worcester, MA, USA
Email:srikanthp@wpi.edu
Research Advisor: Professor Alexander M. Wyglinski

Abstract—In this presentation, we discuss some preliminary results pertaining to an ongoing project on the development of an adaptive signal processing solution combining antenna subset selection, spectral subtraction, and blind source separation in order to mitigate the impact of both wideband jamming and co-site interference by extracting individual transmissions from multiple intercepted mixtures of wireless signals.

I. INTRODUCTION

With the increasing volume of wireless traffic that theatre operations require, the probability of transmissions interfering with each other is steadily growing to the point that new techniques need to be employed. Furthermore, to combat remotely operated improvised explosive devices, many ground convoys transmit high-power broadband jamming signals, which blocks both hostile as well as friendly communications. We aim to devise, implement, and evaluate an adaptive signal processing software solution for mitigating the effects of both intentional and unintentional jamming (including wideband jamming) via the combination of antenna subset selection, spectral subtraction, and blind source separation (BSS) techniques in order to extract specific transmissions from a mixture of intercepted wireless signals. The goal of our proposed solution, called BLInd Spectrum Separation (BLISS), is to enable reliable, high throughput, and robust end-to-end wireless communications in the support of all Department of Navy missions, especially high capacity multimedia (voice, data, imagery) transmissions.

II. TECHNICAL APPROACH

The BLISS solution (see Figure 1) integrates three well-known adaptive signal processing algorithms found in the open literature, namely: antenna subset selection, spectral subtraction, and blind source separation (BSS). Each of these algorithms is employed within the BLISS framework in order to enable the process of extracting individual transmissions intercepted from several mixtures of wireless signals. Although BSS can readily extract transmissions under ideal conditions, the BLISS framework will be deployed in challenging operational scenarios which could significantly degrade the performance of the communication system, or even cause it to fail. Hence, the other two algorithms, i.e., antenna subset selection and spectral subtraction, are employed to make the BSS process more robust in challenging operational conditions. Once the implementation of each of these techniques is performed, the next step is their integration into a seamless BLISS framework. Finally, validation of the novel framework is performed via computer simulations over a range of different operating scenarios.

Blind Signal Separation

Blind signal separation is the task of separating signals when only their mixtures are observed. The process is often termed “blind”, with the understanding that both source signals and mixing procedure are unknown. The mathematical model that describes the observed mixtures as a function of independent components is as follows:

\[ x = Hs + n \]  

In reality, the specific mixing model is the paramount piece of prior information required, and in many scenarios even knowledge of certain source statistics is necessary. Under this setting, the channel may generally be construed as a linear time-invariant (LTI) system, though there is some activity occurring with nonlinear mixing models. Algorithms which rely on this concept, the separation-independence equivalence, may be classed as those performing Independent Component Analysis (ICA). The problem of BSS is then reduced to a mathematical optimization problem, of which several approaches exist, e.g., kurtosis, mutual information, cross power-spectra, entropy, log-likelihood [1]. For instance, the optimization problem based on kurtosis in two-dimensions can...
be formulated as the minimization of
\[ F(q) = q_1^2 + q_2^2 \] (2)
over \( q_1^2 + q_2^2 = 1 \) where the vector, \( q \) is a function of the mixing matrix. The BSS problem for the case of an idealized 2 × 2 BPSK system is illustrated in Figure 2.

In this work, we will devise an unmixing system designed to separate wireless transmissions operating in the presence of jamming fields, especially those with wideband characteristics. Furthermore, we will investigate technical challenges that will be potentially encountered by the BLISS-enabled platform and devise solutions for them.

**Antenna Subset Selection**

Multiple-input multiple-output (MIMO) wireless solutions i.e, the use of multiple antennas at transmitter and receiver, has emerged as a cost-effective technology in making high data rates a reality. This is due to the inefficiency associated with the implementation of a brute force approach of achieving high data rates using a single transmit-single receive antenna system which is theoretically feasible provided that the product of the bandwidth (Hertz) and spectral efficiency (bits per second per Hertz) matches the desired data rate. However, deploying transceivers with multiple antennas requires multiple RF chains that are typically very expensive. Therefore, there is considerable incentive for low-cost, low-complexity techniques with the benefits of multiple antennas. Optimal antenna subset selection is one such technique. A selection of antenna elements, which are typically much cheaper than RF chains, is made available at the transmitter and/or receiver. Transmission/reception is performed through the optimal subset. Several criterion have been proposed in order to arrive at the optimal subset under a variety of operating conditions [2]–[5].

**Spectral Subtraction**

Spectral subtraction is a technique which has been traditionally employed for enhancing speech corrupted by broadband noise. In the context of speech enhancement, the technique involves subtracting an estimate of the noise power spectrum from the speech power spectrum, setting negative differences to zero, recombining the new power spectrum with the original phase, and then reconstructing the time waveform [6]–[8]. In our implementation of the spectral subtraction, instead of subtracting noise, we subtract the power spectral densities of known signals from those that need further processing.

**Integrated Framework**

As for the overall framework that combines antenna subset selection, spectral subtraction and blind signal separation, by applying an optimal antenna subset selection algorithm, the signal at the end of the transmitter passes through the RF chain corresponding to a subset of the available antennas and experiences the distortion caused by the wireless channel. This wireless channel is modeled as a mixing matrix as shown in Figure 2. At the receiver end, a spectral subtraction technique is applied to subtract all of the known signals. After the known signals are subtracted, the problem is essentially a blind signal separation problem. We would have a set of N signals that are received over M sensors. Nongaussianity of noise is assumed and by applying a BLISS technique especially tailored to the case of jamming-resistant communications, the signals are received.

**References**

System Modeling, Simulation, and Design for Beamforming Using Simulink HDL Coder

Yanjie Peng
Department of Electrical and Computer Engineering
Worcester Polytechnic Institute, Worcester, MA, USA
Email: yjpeng@WPI.EDU

Research Advisor: Professor Xinming Huang and Professor Andrew G. Klein

Abstract—Beamforming is a signal processing technique used in sensor arrays for directional signal transmission or reception. It has found numerous applications in radar, sonar, seismology, wireless communications, speech, acoustics, and biomedicine. Adaptive beamforming is used to detect and estimate the signal-of-interest at the output of a sensor array by means of data-adaptive spatial filtering and interference rejection. We focus on the study of using Matlab/Simulink tools for model simulation and hardware design of the traditional linear array beamforming using QR-RLS (QR-decomposition-based recursive least-squares) algorithm. We present the details on the theoretical algorithm, corresponding simulation model, and the FPGA result of the hardware generated by Simulink HDL Coder.

I. INTRODUCTION

In the recent decade, several technologies have been proposed to achieve MIMO (multiple-input and multiple-output) from a conventional SISO (single-input and single-output) system. Among them beamforming [1] is a signal processing technique used in sensor arrays for directional signal transmission or reception. It has found numerous applications in radar, sonar, seismology, wireless communications, speech, acoustics, and biomedicine. Adaptive beamforming is used to detect and estimate the signal-of-interest at the output of a sensor array by means of data-adaptive spatial filtering and interference rejection.

We present the design of the traditional linear array beamforming based on QR-RLS (QR-decomposition-based recursive least-squares) algorithm [1] using Simulink HDL Coder [5]. The theoretical algorithm is discussed in Section II. System modeling using Simulink HDL Coder is introduced in Section III, followed by hardware implementation result in Section IV. Conclusions and future work are given in Section V.

II. LINEAR ARRAY ADAPTIVE BEAMFORMING

The block diagram of the traditional linear array beamforming (M sensors) is presented in Fig. 1. \( s(\phi) \) is the steering vector containing the information of the direction of the target signal. \( w_1^*(n), w_2^*(n), \ldots, w_M^*(n) \) is the adjustable complex weight for sensor 1,2,..,M, respectively.

The requirements for MVDR (minimum-variance distortion-less response) beamforming is: 1) protecting the target signal, and 2) minimizing the interference. The QR-RLS algorithm is a numerically stable solution to MVDR problem. The algorithm is shown in Algorithm 1. The output of beamformer is \( e(n) = \frac{e'(n)}{||a(n)||^2} \).

III. SIMULATION MODEL

The systolic array structure [3] is very suitable for implementing the QR-RLS algorithm. The simulation model of systolic array for 3-sensor beamforming build by Simulink HDL Coder is shown in Figure 2. The array contains two types of processing cells: boundary cells (blue) and internal cells (red). The boundary cells perform the vectoring operation on received signal to form rotation angles used by internal cells. The internal cells perform Givens rotations of the received signal by the angles passed from the boundary cells. Simulation result given 50 snapshots is shown in Fig. 3.

IV. HARDWARE IMPLEMENTATION RESULTS

After fixed point simulation, we can obtain synthesizable verilog and/or hdl code directly by HDL Coder. The FPGA result is shown in Table 2.

V. CONCLUSIONS AND FUTURE WORK

We present the design of the traditional linear array beamforming based on QR-RLS (QR-decomposition-based recursive least-squares) algorithm using Simulink HDL Coder. The
Algorithm 1 QR-RLS Algorithm

- Initial condition $\Phi^{1/2}(0) = I$, and $a(n) = s(\phi)$
- For $n = 1, 2, \ldots$, compute

$$
\begin{bmatrix}
\Phi^{1/2}(n-1) & u(n) \\
\Phi^H(n-1) & 0
\end{bmatrix}
\begin{bmatrix}
\Theta(n) = \\
\Phi^{1/2}(n) & 0
\end{bmatrix}
\begin{bmatrix}
\Phi^H(n)
\end{bmatrix}
\begin{bmatrix}
x(n-1) & -e'(n) \gamma^{-1/2}(n)
\end{bmatrix}
$$

where $s(\phi) = [1, e^{-j \phi}, \ldots, e^{-j(M-1)\phi}]^T$ is the steering vector, $u(n)$ is the received data at time $n$, and $\Theta(n)$ denotes Givens rotation.

Figure 2. Systolic array for 3-sensor beamforming

Figure 3. Simulation result of 3-sensor beamforming

Table I

<table>
<thead>
<tr>
<th>Product Version</th>
<th>Xilinx ISE V11.2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Target Device</td>
<td>Xilinx Virtex5 XC5VLX330</td>
</tr>
<tr>
<td>Number of Slice Registers</td>
<td>1,824 out of 207,360 (1%)</td>
</tr>
<tr>
<td>Number of Slice LUTs</td>
<td>5,355 out of 207,360 (2%)</td>
</tr>
<tr>
<td>Number of DSP48Es</td>
<td>103 out of 192 (53%)</td>
</tr>
<tr>
<td>Maximum Speed</td>
<td>72.6 MHz</td>
</tr>
</tbody>
</table>

FPGA RESULT FOR 3-SENSOR BEAMFORMING

References

Wind Energy Conversion System Operation as Power Generator and Active Filter

Grazia Todeschini
Department of Electrical and Computer Engineering
Worcester Polytechnic Institute, Worcester, MA, USA
Email: grazia@wpi.edu

Abstract—This research deals with the performance of a Wind Energy Conversion System (WECS) operating as power generator and Active Filter simultaneously. As a power generator, the WECS converts wind energy into electric energy; as an AF, it sinks the harmonic currents injected by Non-Linear Loads (NLLs) connected at the same feeder.

Three control systems are developed to ensure the described operation and a specific study regarding the compensation of the triplen harmonics is carried out. WECS derating and voltage distortion are expressed as function of the harmonic currents injected by the system.

The WECS performance as generator and AF has been studied for steady-state analysis, fast transients (voltage variations) and slow transients (wind speed variations). Simulation results of a typical plant show that the proposed control systems allow operating the WECS as power generator and AF both during steady state and transient operation; the described operation causes power loss increase and voltage distortion that require a conservative choice of the WECS components and require system derating.

I. INTRODUCTION

Distributed generation represents an answer to the growing demand of electric energy and to the increasing environmental concerns. Among alternative sources, wind energy is one of the most promising technologies. In the developed countries, wind capacity has been growing at a rate of 20% to 30% a year over the last decade [1]. The rapid growth of the wind industry is due to different factors, including [2]: supporting government policies, advances in wind power technology that caused cost reduction and performance improvement, environmental concerns and the deregulation process.

Given the advances in power electronics and control, applications such as reactive power compensation, static transfer switches, energy storage, variable-speed generation, voltage control and dynamic reactive power support are commonly found in modern wind power plants [3]: these applications are known as ‘ancillary services’. The present research presented in this paper investigate one of the ancillary services: active filter operation.

II. THE STUDIED SYSTEM

The studied system is shown in Fig. 1. The Doubly-Fed Induction Generator (DFIG) stator terminals are connected to the Point of Common Coupling (PCC) through a transformer and a feeder, represented by the equivalent resistance and inductance $R_c$ and $L_c$. The feeder that connects the Non-Linear Load (NLL) to the PCC is represented by the equivalent resistance and inductance $R_h$ and $L_h$. The DFIG rotor is supplied by two back-to-back connected converters: the rotor side converter (RSC) and the line side converter (LSC). The feeder that connects the LSC to the PCC has the equivalent resistance and inductance $R_L$ and $L_L$. The RSC and LSC power switches are driven by means of Pulse Width Modulation (PWM). The control system design is performed in an equivalent $dq0$ domain obtained by applying Park transformation to the three-phase variables [4].

III. COMPENSATION BY MEANS OF COMBINED MODULATION

As proved in [5], the most effective AF strategy uses both power converters and the DFIG by combining compensation by means of RSC and LSC modulation. This strategy is named ‘combined modulation’ (CM). The aim of this method is to distribute the harmonic power flow and therefore the power loss increase among the DFIG, the RSC and the LSC.

When compensation by means of CM is applied, the zero-sequence harmonics injection is obtained by modulating the LSC, and compensation of the symmetrical harmonic components is obtained by modulating the RSC. In terms of $dq0$ components, the reference harmonic currents for the LSC and the RSC control systems are as follows:

$$i_{dL,ref} = -i_{0h}$$
$$i_{ds,ref} = -i_{dh}$$
$$i_{qs,ref} = -i_{qh}$$

where $i_{dh}, i_{qh}, i_{0h}$ are the $dq0$ currents measured at the NLL terminals.

A. Steady-state analysis

Steady-state analysis allows verifying the validity of the proposed control system and quantifying its effects.

The current and voltage THD (Total Harmonic Distortion) are measured at the PCC for a case study, before and after
harmonic compensation is implemented. Tab. I lists the results and shows a significant THD reduction when harmonic compensation is applied.

Two effects of the proposed application on the WECS operation are also identified:

1) power loss increase and consequent temperature rise; under certain operating conditions, derating of the WECS is necessary to limit the power loss and the devices temperature.

2) voltage distortion due to the harmonic current flow and harmonic voltage drop on the line impedances. It results that the measured peak voltage at the stator terminals is above the rated value and requires conservative insulation design.

B. Transient analysis

The system transient response is investigated for two types of disturbances: voltage variation (fast transients) and wind speed variation (slow transients). An extensive number of cases have been studied and the results are summarized as follows:

- harmonic compensation is not affected by transient operation, since the reference harmonic currents are measured at the NLL terminals;
- derating helps reducing severity of current during the transients because it causes a reduction of the fundamental current amplitude;
- LVRT ability is verified: the WECS continues to provide power and is not damaged according to the requirements presented in [6];
- wind speed variation introduces low frequency current disturbances, due to the mechanical system response.

IV. CONCLUSION

The simulation results show that the WECS can be used simultaneously as Active Filter and Power Generator. Steady-state analysis shows that to safely operate the system at high wind speeds, derating and a conservative choice of the system components are necessary. Transient phenomena have no effects on the WECS ability to inject harmonic currents.

REFERENCES

Stress Field Calculation for Quasi-static Ultrasound Elastography via Force Sensor Integration and SLE

Lili Yuan, Ultrasound Laboratory, ECE Department
Advisor: Professor Peder C. Pedersen

Abstract—Most current elastography methods remain qualitative or display only strain information due to the inexact internal stress distribution. A more accurate result may be obtained by employing force sensors to gauge the applied vertical compression force on skin surface through the ultrasound transducer. A computationally efficient superposition algorithm based on Love’s closed-form equation (SLE) is presented. It performs analytical calculation of the 3D dynamic stress field inside a soft tissue phantom with non-free boundary conditions under a non-uniformly stressed rectangular linear array transducer, using the pressure on each element of the contact surface. The validity of the SLE method was tested by comparison to the stress field calculated with Finite Element Analysis (FEA). For the region of interest (directly underneath the ultrasound transducer), the SLE provides a sufficiently precise solution and can be exploited for absolute Young’s modulus reconstruction in real time.

I. INTRODUCTION

Ultrasound elastography is a recent technology for non-invasively imaging diseased arteries, detecting prostate tumors and categorizing breast lesions [1]. Well developed quasi-static free-hand techniques mostly involve only tissue strain. However, the stiffness may be dependent on the magnitude and direction of the force applied, especially if the linear stress-strain region (typically 1%-2%) is exceeded [2]. Inner stress as the response of external force is also required in addition to the strain.

Inexpensive, ultra-thin, flexible piezoresistive force sensors (Tekscan, MA) were utilized to quantitatively measure the contact force. The force sensor output was then integrated into the SonixRP (Ultrasoundix, Vancouver, Canada) strain imaging system. 3D stress field was achieved by means of the computed pressure on each force plate element and SLE algorithm.

II. FORCE SENSOR INTEGRATION

A force plate with a slot for the ultrasound transducer was designed, as shown in Fig. 1. Force sensors were mounted beneath the force plate and connected to the signal amplification, noise filtering, and data acquisition system. The force plate was decomposed into small rectangular elements with each individual constant pressure. Contact pressure on every sub-region with a sensor was estimated by measured force divided by sensing area. For the remaining sub-regions without sensors, pressure was obtained by bilinear interpolation.

A combined display of strain imaging and forcing function was implemented. Figure 2 illustrates the outcome with agar-based phantom. Force versus time was used to depict external force magnitude and loading frequency.

III. SUPERPOSITION ALGORITHM BASED ON LOVE’S FORMULA

SLE was introduced for analytical computation of 3D stress field under the assumption of a homogeneous, isotropic, semi-infinite, elastic solid material, stressed non-uniformly by rectangular compressor. The stress at each field point is the overall contribution from every force plate element under individual constant pressure.

IV. RESULTS

A phantom (120mm × 60mm × 40mm, Young’s modulus 5 × 10^4 Pa, Poisson’s ratio 0.42, density 1040 kg/m^3) and

![Fig. 1. Bottom view of linear array rectangular transducer with size of 60 mm × 20mm, force plate geometry a × b, partitioned into small element with l × w.](image)

![Fig. 2. Front view of sensor integration into strain image system, the upper image is strain image superimposed on B-mode with blue and red color respectively representing softer and harder material.](image)

![Fig. 3. Geometry of the ith force element with center and nodes coordinates Expression for the distribution of vertical stress component Szz at field point (x,y,z) is Szz(x,y,z) = ∑n=1m ∫x,y,z 1/2π (dw/dz) - y (dy/dz) (1) where, x0 and y0 denote the number of force elements in x and y direction, V is the Newtonian potential, and dw/dz and dy/dz are calculated by replacing contact area geometry in corresponding equations as x0 = m0 + n0 and x0 = n0 + l0. Elaborate description of these formulas omitted here for saving space can be found in [3].](image)
Coaxial force plate (108 mm × 34 mm) were simulated for 3D stress distribution calculation using both SLE and FEA (COMSOL, Sweden).

Cross-sections coincident with transducer image plane from the two methods were compared to evaluate the accuracy of SLE for stress estimation in visco-elastic material with fixed boundary.

A. Stress Field Calculation by SLE

Stress distribution, due to uniform pressure 1000 Pa, along the axis of compressor decays with depth in a semi-infinite medium, presented in Fig. 4. The counterpart by FEA with the same tendency was described in [4].

Fig. 4. Stress field cross section due to uniform vertical compression by SLE.

As an initial experiment, two force sensors were allocated at two ends of the force plate and the contact surface were divided into three equal-size rectangle, where the pressure are 1000 Pa, 1200 Pa, and 1400 Pa in turn along x axis. Uneven distributed stress closer to surface and gradual reduction with depth is shown in Fig. 5.

Fig. 5. Stress field cross-section for non-uniform normal compression by SLE

B. SLE Performance Evaluation

Stress results from FEA with fixed bottom and prescribed-displacement surrounding boundary (only displacement in z direction is allowed) is presented in Fig. 6, followed by the normalized difference image (Fig. 7). SLE is far more efficient than FEA (9.2 × 10^{-6} s/element vs 1.07 × 10^{-2} s/element ) and less than 5.62% deviation from FEA from surface to depth at 22.3 mm except around the edges.

Fig. 6. 2D stress for non-uniform normal compression using FEA

Fig. 7. Normalized stress difference between SLE and FEA for variable pressure

The mismatch that exists because of gradient of pressure variation, plate and phantom geometry, mesh fineness and interpolation method for FEA simulation is not mentioned in detail due to space limitation.

V. CONCLUSION

By virtue of force sensors and SLE, fast 3D dynamic stress field calculations are achievable, which can be combined with strain to obtain real time quantitative elastography for elastic solid material, therefore improving the quality of tumor and cancer diagnosis.

ACKNOWLEDGEMENT

The support from the Telemedicine and Advanced Technology Research Center are greatly appreciated. The authors also thank research IT support Siamak Najafi for providing COMSOL software for free and thank lab manager Frederick Huston for supplying sensor calibration equipments.

REFERENCES


