Blind Beamforming for Multipath Wideband Signals

Wei Liu

Communications Research Group
Department of Electronic and Electrical Engineering
University of Sheffield
Sheffield, United Kingdom

Email: w.liu@sheffield.ac.uk
Outline

1. Review of Blind Beamforming

2. Introduction to the Wideband Multipath Problem

3. Proposed Approach Based on Frequency Invariant Transformations (FIT)

4. Simulation Results

5. Conclusions
1. Review of Blind Beamforming - Traditional Beamforming
1. Review of Blind Beamforming - Traditional Beamforming
1. Review of Blind Beamforming - Traditional Beamforming
1. Review of Blind Beamforming - Traditional Beamforming

- Each of the received sensor signals is a mixture of all impinging signals.

- Normally we assume that the array geometry and the direction of the signal of interest are known to the system or can be estimated in some way.

- The aim of beamforming is to suppress the interfering signals in order to recover the signal of interest at the output.

- There is a similar problem in blind source separation and a model for blind signal separation is shown in the next slide.
1. Review of Blind Beamforming - Blind Source Separation
1. Review of Blind Beamforming - Blind Source Separation

- The mixing process can be either linear or nonlinear. The linear case can be further divided into two categories:
  - Instantaneous mixing: the mixtures are modeled as weighted sums of individual sources without dispersion or time delay, given by

\[
x[n] = A s[n] + v[n],
\]

(1)

- Convolutional mixing: \(x[n]\) are sums of filtered versions of the sources:

\[
x[n] = \sum_{i=-\infty}^{+\infty} A_i s[n - i] + v[n],
\]

(2)

where \(A_i\) is the matrix impulse response of the linear time invariant multichannel mixing filter.
- Applying blind source separation techniques to the traditional beamforming problem, we then have a blind beamforming system. The advantage with blind beamforming is that we do not need to know the direction of arrival of any signals and the array geometry can be arbitrary and unknown to the system either.
2. Introduction to the Wideband Multipath Problem (1)

- A linear array with $M$ sensors is shown in the following. Suppose $s_l(t)$, $l = 0, \ldots, L - 1$ are the $L$ impinging plane-wave signals that are received at the zero-th sensor. Then the signal received at the $m$-th sensor will be $x_m(t) = \sum_{l=0}^{L-1} s_l(t - \tau_{m,l})$, where $\tau_{m,l}$ is the delay from the zero-th to the $m$-th sensor.

- For convenience, we consider the discrete form of the signals and we then have $s_l[n] = s_l(nT)$, $l = 0, \ldots, L - 1$, and $x_m[n] = x_m(nT)$, $m = 0, \ldots, M - 1$, where $T$ is the sampling period.
2. Introduction to the Wideband Multipath Problem (2)

- A general wideband beamforming structure with tapped delay-line (TDL) based processing:
2. Introduction to the Wideband Multipath Problem (3)

- In this structure, each of the received array signals is processed by a TDL and then added together to form the beamformer output $y[n]$

$$y[n] = w^T x, \quad (3)$$

where $w$ is the coefficients vector and $x$ the vector holding all of the received data samples along the tapped delay-lines.

- Without loss of generality, assume that $s_0(t)$ is the desired signal and $s_1(t)$ is a multipath version of $s_0(t)$ given by $s_1(t) = \alpha s_0(t - \delta T)$, where $\alpha$ is a scalar and $\delta T$ is the corresponding delay. (Note the discussion and the proposed scheme can be extended to the case with more than one multipath interferences straightforwardly.) All of the other $L - 2$ signals are uncorrelated interferences.

- For adaptive beamforming, as an example, we consider the linearly constrained minimum variance (LCMV) beamformer

$$\min_w w^H R_{xx} w \quad \text{subject to} \quad C^H w = f, \quad (4)$$

where $R_{xx}$ is the covariance matrix of observed array data in $x$, $C$ is the constraint matrix and $f$ is the response vector.
2. Introduction to the Wideband Multipath Problem (4)

- The constraint will ensure that no matter how to adjust the array coefficients, the resulting beamformer will have the desired response set out by the constraint equation $C^H w = f$.

- During the minimisation of the output variance, the uncorrelated interferences will be suppressed effectively. However, for the correlated interference $s_1(t)$, it will try to cancel at least part of the desired signal in the output to reduce the overall output signal variance.

- Suppose the impulse response of the beamformer to the desired signal $s_0(t)$ from the direction $\theta_0$ is simply a delay $T_1$ and to $s_1(t)$ from $\theta_1$ is given by $h_1(t)$ i.e.:

  - At the beamformer output, the component corresponding to $s_0(t)$ is $s_0(t - T_1)$ and that corresponding to $s_1(t)$ is $\alpha s_0(t - \delta T) * h_1(t)$.

  - When $h_1(t)$ has a magnitude response $1/\alpha$ and a phase response $-(T_1 - \delta T)\omega$, the desired signal will be canceled completely.
3. Proposed Approach Based on Frequency Invariant Transformations (FIT) (1)

- Frequency invariant beamforming is a technique for wideband array design to form a response only as a function of the direction of arrival of the impinging signals, independent of the signal frequency. An example:
3. Proposed Approach Based on FIT (2)

- The proposed blind beamforming structure for multipath signals is shown in the following, where $x[n]$ is the received array signal vector at time $n$ and each of the blocks labeled as $\text{FIB}_i$, $i = 0, 1, \ldots, N - 1$ represents a frequency invariant beamformer with a response $P_i(\theta)$.
- Each of the $N$ FIBs has its main beam pointing to different directions and together they cover all of the possible directions of interest.

![Diagram of proposed approach]

$\begin{align*}
\text{FIB}_0 & : b_0[n] \times w_0 \\
\text{FIB}_i & : b_i[n] \times w_i \\
\vdots & \\
\text{FIB}_{N-1} & : b_{N-1}[n] \times w_{N-1}
\end{align*}$
3. Proposed Approach Based on FIT (3)

- The FIB outputs $b_i[n], i = 0, 1, \ldots, N - 1$ can be expressed as a weighted sum of the impinging signals, given by

$$b_i[n] = p_i \cdot s[n] \quad \text{with} \quad \begin{cases} p_i = \left[ P_i(\theta_0) \ P_i(\theta_1) \ \cdots \ P_i(\theta_{L-1}) \right] \\ s[n] = \left[ s_0[n] \ s_1[n] \ \cdots \ s_{L-1}[n] \right]^T \end{cases} \quad (5)$$

- $b_i[n]$ is exactly a weighted sum or an instantaneous mixture of the original $L$ source signals. Then we can apply a simple instantaneous BSS algorithm to the $N$ outputs to extract the desired signal. A simple version of BSS is called blind source extraction (BSE), where the desired signal is extracted by applying the set of coefficients $w_i$, $i = 0, 1, \ldots, N - 1$. The output $y[n]$ is given by

$$y[n] = \sum_{i=0}^{N-1} w_i b_i[n] \quad (7)$$
3. Proposed Approach Based on FIT (4)

- The component \( b_{i,l}[n] \) corresponding to \( s_l[n] \) in the output \( b_i[n] \) of FIB\(_i\) will be
  \[ b_{i,l}[n] = P_i(\theta_l) s_l[n] \]

- Then, for the desired signal \( s_0[n] \), its corresponding component \( y_0[n] \) in the output \( y[n] \) will be
  \[ y_0[n] = \left( \sum_{i=0}^{N-1} P_i(\theta_0) w_i \right) s_0[n] \]  
  (8)

- For the component \( y_1[n] \) in the output \( y[n] \) corresponding to \( s_1[n] \), the scaled delayed version (multipath) of \( s_0[n] \), it is
  \[ y_1[n] = \left( \sum_{i=0}^{N-1} P_i(\theta_1) w_i \right) s_1[n] = \alpha \left( \sum_{i=0}^{N-1} P_i(\theta_1) w_i \right) s_0(nT - \delta T) \]  
  (9)

- Then the signal of interest \( y_s[n] \) in the output \( y[n] \) will be the sum of them
  \[ y_s[n] = \left( \sum_{i=0}^{N-1} P_i(\theta_0) w_i \right) s_0[n] + \alpha \left( \sum_{i=0}^{N-1} P_i(\theta_1) w_i \right) s_0(nT - \delta T) \]  
  (10)
3. Proposed Approach Based on FIT (5)

• The BSS/BSE algorithms are based on some assumptions on the statistical properties of the source signals. For example, if the original source signals are independent of each other with at most one Gaussian signal, we can separate the original signals based on the principle that the separated signals are as independent of each other as possible.

• For the multipath case when the source signal $s_1[n]$ is a delayed version of $s_0[n]$, we will not be able to separate them from each other. However since both of them are independent of the remaining signals, as long as the number of mixtures $N$ is not smaller than the source signal number $L$, we will be able to separate them from the remaining signals by finding an appropriate linear combination $w$ of the mixtures $b[n]$.

• As a result, in the ideal case, we will only have $y_s[n]$ left in the output $y[n]$ and it is a simple filtered version of the desired signal $s_0(nT)$.

• Given $y_s[n]$, the original signal $s_0[n]$ can be recovered by some appropriate deconvolution methods.
4. Simulation Results (1)

- Simulations are based on a uniform linear array with \( M = 17 \) sensors.

- Desired signal: broadside \( \theta_0 = 0 \); two uncorrelated interfering signals: \( \theta = 60^\circ \) and \( -50^\circ \); SIR=0 dB; Bandwidth: \( \Omega \in [0.45\pi, \pi] \).

- Two correlated multipath signals, which are delayed versions of the desired signal by 10 and 20 samples and arrive from the directions \(-70^\circ\) and \(30^\circ\), respectively. Both have a magnitude scaled by 0.5 compared to the desired signal.

- The original LCMV beamformer has dimensions of \( 17 \times 80 \). For the spatial-smoothing based LCMV beamformer, the sub-array size is 6 and in total there are \( 17 - 6 + 1 = 12 \) sub-arrays, whose outputs are added together to form a new LCMV beamformer with dimensions of \( 6 \times 80 \).

- For the proposed scheme, 5 frequency invariant beams are designed, each of them with dimensions \( 17 \times 80 \), and their desired responses are given in the next slide.
4. Simulation Results (2)

- The desired response of the five FIBs:
4. Simulations Results (3)

- The output SINRs of the original LCMV beamformer, the spatial-smoothing based LCMV beamformer and the FIB-based beamformer
4. Simulation Results (4)

- The proposed FIB based blind beamformer can extract the desired signal successfully, with its output SINR increasing steadily with respect to a reducing input noise level.

- On the other hand, the traditional LCMV beamformer has failed to pass through the signal of interest at its output and its output SINR stays at about 8 dB with a tiny fluctuation with respect to the input SNR level.

- The spatial-smoothing method can improve the output SINR of the traditional LCMV beamformer, but the improvement is very limited and not as significant as the FIB based one.
5. Conclusions

- A blind beamforming approach for multipath wideband signals based on a frequency invariant transformation has been proposed.

- In this approach, the received array signals are first processed by a frequency invariant beamforming network and then an appropriate instantaneous BSS/BSE algorithm is applied to recover the desired signal.

- Simulation results show that a significantly improved output SINR has been achieved by the proposed scheme and it can deal with the correlated wideband interference problem effectively.
THANK YOU!