MIMO MEASUREMENTS OF COMMUNICATION SIGNALS AND APPLICATION OF BLIND SOURCE SEPARATION

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ABSTRACT

In this paper we present a flexible multiple input multiple output (MIMO) measurement system for communication signals in the 2.4 GHz band. We present some measurements of the digital to digital transmission channel which includes all impairments of the hardware realization.

Using this system we perform a multi-layer transmission of communication signals. At the receiver side we use blind source separation (BSS) techniques as frontend processing to avoid estimation and synchronization problems. In order to improve the estimation of the channel and the symbol detection, we propose an iterative approach, which uses the knowledge of the finite symbol alphabet but does not need additional training data.

1. INTRODUCTION

Multiple input multiple output (MIMO) systems are currently under discussion for communication systems because of their ability to exploit the capacity of a spatial channel. Recently, powerful detection algorithms have been developed. These algorithms need a suitable estimation of the transmission channel. In most cases the channel estimation is obtained by including a pilot sequence in the data stream. However, this will lead to a loss of spectral efficiency. Therefore we will present a new scheme that combines blind source separation techniques with an efficient detection algorithm in an iterative way. This will lead to an overall blind detection of the transmitted symbols.

The remainder of the paper is organized as follows: In the current section we will introduce briefly our measurement system for flexible handling of MIMO transmissions. In section 2 we will present some measurements on the frequency response of the MIMO channel under consideration. Section 3 introduces a setup for using BSS techniques in a communication environment. Some measurements illustrate the feasibility of this approach. In section 4 we introduce an iteration scheme in order to utilize the finite symbol alphabet. The measurements include an estimation of the signal noise ratio (SNR) to illustrate the advantages of such an approach. A summary and concluding remarks can be found in section 5.

1.1. Motivation and System Description

When simulating transmission systems many estimation problems are normally ignored or values are taken as ideally know. Several assumption have to be made in order to model the transmission channel. Finally a simulation only approach may lead to results that cannot be reproduced in a hardware implementation. Therefore we build a flexible hardware demonstrator for MIMO communication setups. This simulator introduces different impairments to the signal and demands the inclusion of many estimation tasks.

Our general concept is to have a real world transmission, but an offline processing of the signal in order to investigate algorithms that cannot be implemented in realtime today but in the near future. This offline processing approach gives us the flexibility to investigate most MIMO setups currently under discussion. Our 2.4 GHz demonstrator is based on a direct conversion transmitter and receiver concept. The transmitter and the receiver as well consit of 8 modules that are synchronized according to their local oscillators (LO) and digital control. The transmitter sends an arbitrary periodically repeated signal from its memory (up to $8 \times 512\mathrm{k}$ samples). The receiver takes a snapshot of the transmitted signal with twice the period length of the transmitted signal. Sampling frequencies up to 50 MHz are possible. The received signal is transfered to a PC using an USB interface and can be processed by a simulation tool of choice.

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2. FREQUENCY RESPONSES

In this section we will present a setup for measuring the frequency response of the MIMO transmission channel. We consider the channel from the digital domain at the transmitter to the digital domain at the receiver, thus including all effects of the system components. We have to emphasize that it is not our intention to do systematic channel measurements.

For measurements we apply a chirp-like signal, whereas only one transmitter is sending at a time, in order to measure the complete matrix of frequency responses (**figure 1**).

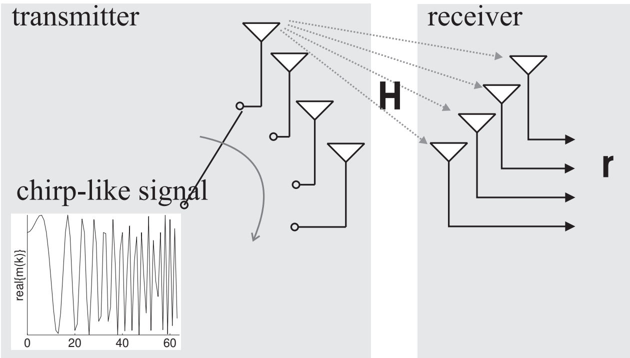

Fig. 1. Multiplexing for channel measurement

This signal is designed in the frequency domain as

$$M(n) = e^{-j\frac{\pi}{N_{\mathrm{DFT}}}n^2} \quad \text{for} \quad n = 0 \dots N_{\mathrm{DFT}} - 1 \quad , \quad (1)$$

because this guarantees an exactly flat magnitude. Processing the IDFT, we get the time domain signal

$$m(k) = \mathrm{IDFT}_{N_{\mathrm{DFT}}}\{M(n)\} \quad (2)$$

which is inherently periodic. We exploit this property and send $m(k)$ in a periodic way so that only a coarse synchronization is necessary.

The quadratic phase increment leads to a small crest factor[1] of the signal.

We can measure the frequency response, up to a linear phase uncertainty, by using a fractional part of the received time signal $r(k)$ with N_{DFT} samples and calculating

$$\begin{aligned} R(n) &= \mathrm{DFT}_{N_{\mathrm{DFT}}}\{r(k + k_{\mathrm{offset}})\} \\ & \quad k = 0 \dots N_{\mathrm{DFT}} - 1 \end{aligned} \quad (3)$$

$$H(n) = \frac{R(n)}{M(n)} \quad . \quad (4)$$

Since this method is sensitive regarding the frequency offset, we added a pilot sequence to our measurement frame in order to estimate and correct the offset.

[1] The peak-to-rms voltage ratio of an alternating current (ac) signal

The advantage of this approach is that we only need a coarse synchronization and not a high-precision time reference (like in channel sounding setups).

Therefore the starting position k_{offset} may be slightly inaccurate by k_{shift}. This circular[2] time shift of the starting position will result in a linear phase term, but it does not influence the shape of the magnitude response:

$$\begin{aligned} H_{\mathrm{shift}}(n) &= \frac{\mathrm{DFT}_{N_{\mathrm{DFT}}}\{r(k+k_{\mathrm{offset}}+k_{\mathrm{shift}})\}}{M(n)} \\ &= H(n)e^{j\frac{2\pi}{N_{\mathrm{DFT}}}nk_{\mathrm{shift}}}. \end{aligned} \quad (5)$$

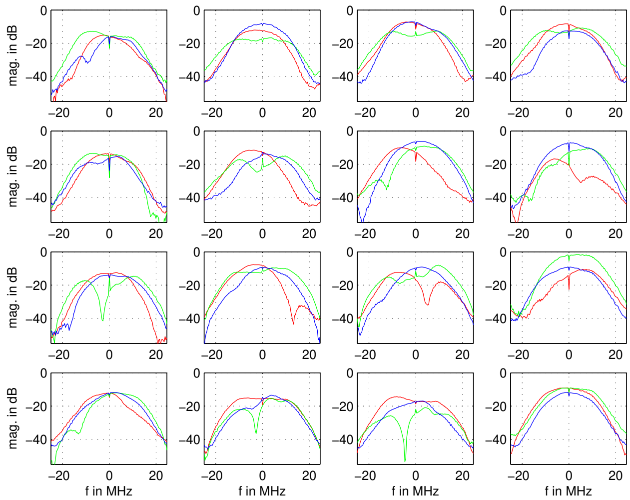

Fig. 2. Frequency responses for a 4x4 setup; a column represents the responses excited by one transmitter

Figure 2 depicts 3 different frequency response measurements using 4 transmit and 4 receive antennas. The measurements were taken from one office room to an adjacent office. Uniform linear arrays (ULAs) with $\lambda/2$ spaced elements are used. The sampling frequency was set to $f_s = 50$ MHz.

The filter influence of our transmissions system, which limit the signal to the 3 dB range of approx. ± 16 MHz, can be seen directly. In addition there are some deep fades in the spectrum that arise from a frequency selective channel. Our measurements revealed, that already a small change of the position may have a strong impact on the frequency response.

3. BLIND SOURCE SEPARATION

Blind source separation (BSS) algorithms are able to separate different signals from a multi sensor setup. The only

[2] Since we are using a periodic repeated signal, we can interpret a time shift as a periodic time shift.

knowledge used to achieve this goal is that the signals should be statistically independent.

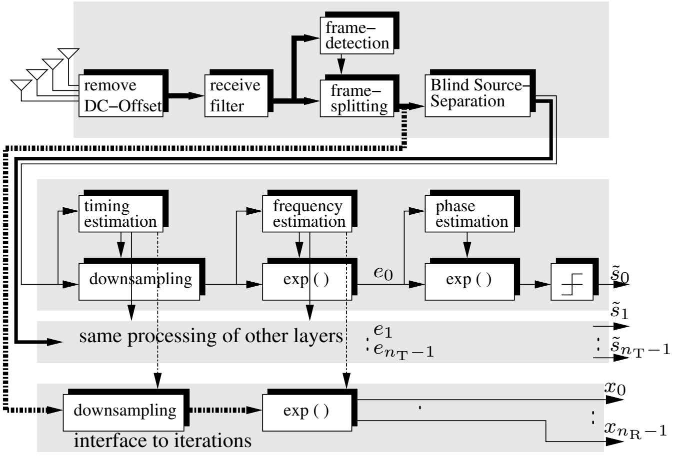

Fig. 3. receiver setup for BSS

To apply source separation techniques in communications we are using the setup depicted in **figure 3**. First of all the DC-offset caused by the direct conversion frontend is removed. After root raised cosine filtering a frame synchronization based on a power detection is carried out. To separate the independent components we can apply a BSS algorithm directly to to the oversampled signal. For this step we choose the JADE [1] algorithm as a spatial only separation approach.

The separation leads to data streams which are processed in the classical way like in single antenna systems. We synchronize to the symbol timing using the method presented in [2]. In order to determine the carrier frequency offset we apply a non-linearity and a frequency estimation.

Measurements were done with a sampling frequency of $f_s = 50$ MHz. The distance between transmitter and receiver was spanning two office rooms. With an oversampling of $w = 8$ we can consider the transmission channel as nearly flat. In order to visualize the successful separation we simultaneously transmit signals with different modulation schemes.

Figure 4 depicts the separation of a BPSK, QPSK, 8PSK and a 16QAM signal sent in parallel and received by four antennas. The signal constellation before separation is obtained by using the timing information estimated after separation. As one can see in figure 4 the signal streams are properly separated.

Based on our experiences we can state that it is practically possible to apply separation algorithms for separation of communication signals in MIMO setups, even if the properties of the modulation schemes are not taken into account. This makes our setup interesting for interference scenarios.

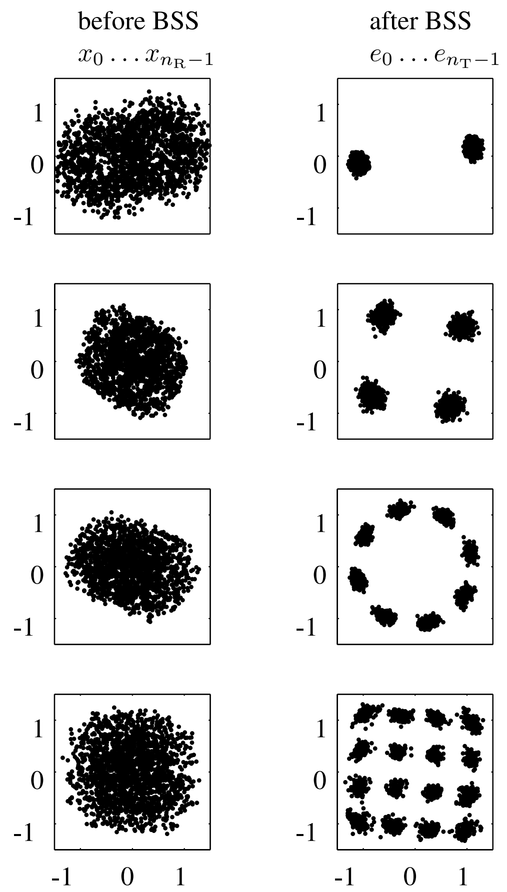

Fig. 4. 4x4 signal constellations before and after BSS

4. BLIND SOURCE SEPARATION WITH ITERATIVE IMPROVEMENT

The general form of BSS approaches do not take into account the finite alphabet of the modulation signals. Therefor we present a combination of the BSS and a powerful detection scheme. As a MIMO detections scheme we choose the VBLAST [3] system.

Figure 5 depicts the idea of the combination. The symbols $\tilde{s}_0 \dots \tilde{s}_{n_T-1}$ detected using the blind source separation (figure 3) are used as reference symbols in order to estimate the spatial channel matrix $\tilde{\mathbf{H}}^{(i)}$. The estimation is performed by taking the pseudo inverse of the previously detected symbols. Then new data symbols $\tilde{s}_0^{(i)} \dots \tilde{s}_{n_T-1}^{(i)}$ will be detected using the VBLAST algorithm with this channel matrix. This new symbols can be used as reference symbols once again for a new channel estimation. These iterations will improve the detection, because the blind source separation can only design a linear filter but the VBLAST algorithm will introduce a successive interference cancellation

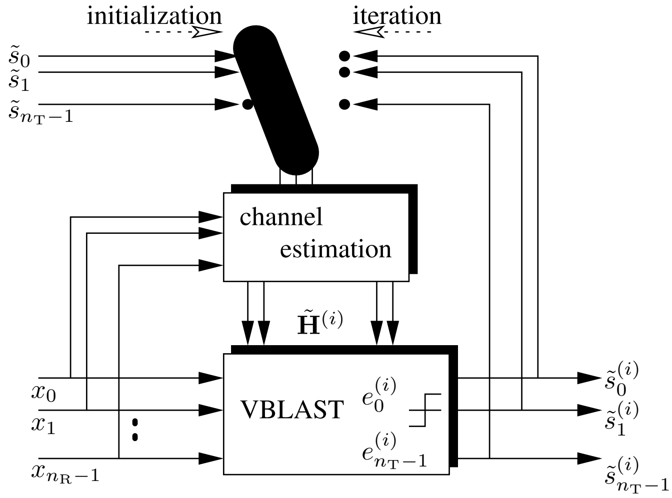

Fig. 5. iterative improvement of channel estimation and detection

that utilizes the knowledge of the finite alphabet.

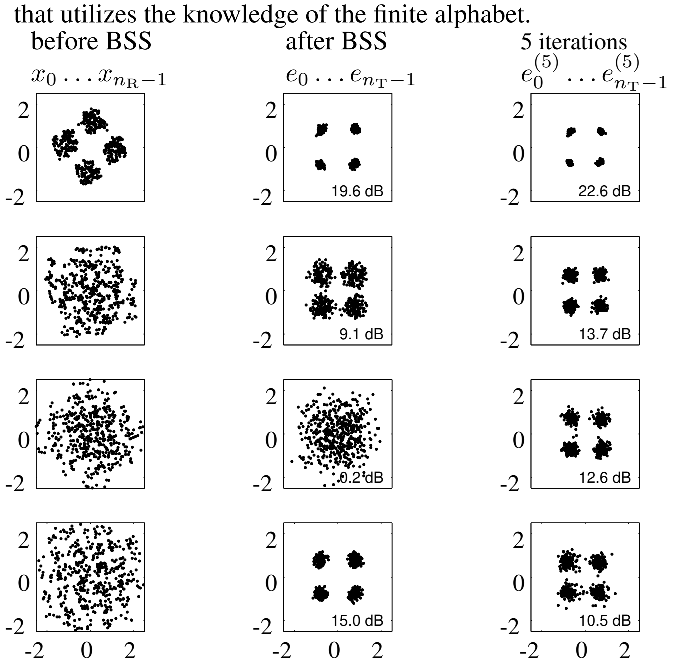

Fig. 6. signal constellations before, after blind source separation and after 10 iterations – estimated SNR

In order to show the feasibility of this approach four QPSK signals are transmitted in parallel. On the receiver side the schemes presented in figure 3 and 5 were used. **Figure 6** depicts the signal constellation before the separation, after the separation and after some iterations of the the scheme presented in figure 5. We have included estimations of the SNR of the symbol constellation before the decision devices [4]. In figure 5 a signal constellation after the BSS consists nearly of noise. Using the proposed iteration scheme even this constellation can be resolved to QPSK. The reason for this is the ranking of the successive interference cancellation in the VBLAST detection. This ranking is based on the estimated channel coefficients that are refined iteratively.
It can be stated that by utilizing the symbol alphabet in an iterative way the SNR can be significantly improved.[3]

5. CONCLUSIONS

In this paper we introduced briefly a measurement system which allows the testing of nearly all MIMO communications setups currently under discussion. Arbitrary signals can be generated and transmitted in realistic scenarios. This requires the solution of different estimation problems.

We presented measurements showing the feasibility of BSS techniques for communications systems under realistic conditions. Our setup utilizes BSS as a robust frontend processing. This gives us the option to use known SISO estimation schemes for frequency and timing estimation. We also presented a new concept to improve the performance of a linear BSS by utilizing the finite alphabet of the modulated signal in an iterative loop.

6. REFERENCES

[1] Jean-François Cardoso and Antoine Souloumiac, "Blind beamforming for non Gaussian signals," *IEE Proceedings-F*, vol. 140, no. 6, pp. 362–370, December 1993.

[2] S.J. Lee, "A New Non-Data-Aided Feedforward Symbol Timing Estimator Using Two Samples per Symbol," *IEEE Communications Letters*, vol. 6, no. 5, pp. 205–207, May 2002.

[3] P. W. Wolniansky, G. J. Foschini, G. D. Golden, and R. A. Valenzuela, "V-BLAST: An Architecture for Realizing Very High Data Rates Over the Rich-Scattering Wireless Channel," *invited paper, Proc. ISSSE-98*, 29. September 1998.

[4] Rolf Matzner, "An SNR Estimation Algorithm for Complex Baseband Signals using Higher Order Statistics," *FACTA UNIVERSITATIS: ELECTRONICS AND ENERGETICS*, vol. 6, no. 1, pp. 41–52, January 1993.

[5] J. Rinas and K.D. Kammeyer, "Comparison of Blind Source Separation under the Application of the Turbo Principle," in *5th International ITG Conference on Source and Channel Coding*, Erlangen, Germany, January 2004, submitted.

[3] In [5] we present some Monte Carlo Simulations concerning the performance of the proposed iterative scheme.