2015 IEEE DIGITAL SIGNAL PROCESSING PROJECT LIST BASED ON MATLAB

1. A Novel Decorrelation Approach For An Advanced Multichannel Acoustic Echo Cancellation System

A multichannel sound reproduction system aims at offering an immersive experience exploiting multiple microphones and loudspeakers. In the case of multichannel acoustic echo cancellation, a suitable solutions for overcoming the wellknown non-uniqueness problem and an appropriate choice of the adaptive algorithm become essential to improve the audio reproduction quality. In this paper, an advanced system is proposed based on the introduction of a multichannel decorrelation solution exploiting the missing-fundamental phenomenon and a combined multiple-input multiple-output

architecture updated by using the multichannel affine projection algorithm. Experimental results proved the effectiveness of the presented framework in terms of objective and subjective measures, providing a suitable solution for echo cancellation.

2. Optimal Factoring of FIR Filters

New insights suggest that the most efficient FIR digital filters can be created by using a scaled sequence of stages, each representing a factor of the filter's transfer function. A crucial capability for building such filters concerns finding the best FIR filter factors, then carefully scaling and sequencing them. The efficiency of the resulting structure depends heavily upon obtaining such optimalfactors. We offer an algorithm to find, scale and sequence optimally factored FIR filters.

3. Partial-Aliasing Correlation Filters

Correlation filters (CFs) are useful tools for detecting and locating signals or objects within a larger signal or scene of interest. Typically, these filters are designed during the training stage without worrying about how the cross-correlation between a test

signal and the designed CF template will be carried out during the testing or use stage. Because of its computational benefits, the Fast Fourier Transform (FFT) algorithmis usually used for performing cross-correlations, leading to circular correlations and aliasing in the resulting correlation outputs. The aliasing effects can be suppressed

by zero-padding, but at the expense of using longer FFTs and thus incurring more computational complexity. In this paper, we present a new approach where CFs are designed to explicitly allow partial aliasing at test time (thus allowing the use of shorter

FFTs). This approach of allowing aliasing in the cross-correlation output and explicitly taking such partial aliasing into accountwhen designing the CF is diametrically opposite to the conventional CF approaches which try to avoid aliasing effects. We demonstrate through numerical results that these new partial-aliasing correlation

filters (PACFs) achieve better recognition performance than conventional CFs when used in block filtering architectures that allow aliasing.

4. A Distributed Arithmetic based Approach for the Implementation of the Sign-LMS Adaptive Filter

A Distributed Arithmetic (DA) based scheme for the implementation of Sign-LMS adaptive filter is presented. DA is an efficient technique for the computation of the dot product of two vectors. This is done by storing the pre-computed partial-products in memories which are then shift-accumulated for the computation of the output. DA can be used for the realization of the finite impulse response (FIR) filters, however, for the realization of the adaptive filters, the partial-products have to be updated from time to time. This is achieved by using a memory which stores the partial-products of the set of recent input samples. The proposed scheme has a convergence performance similar to that of the multiply-and-accumulate (MAC) based implementation. Results show that the throughput of the DAbased implementation is better than the MAC based implementation. Further, it is observed that the throughput is almost a constant with respect to the filter order which makes it more suitable for implementing large filters.

5. AS-band Bitstream Transmitter with Channelized Active Noise Elimination (CANE)

Bitstream Modulated transmitters based on Delta Sigma Modulation have been proved to be promising for efficient amplification of non-constant envelope signal. Such transmitters preserve signal linearity by converting the signal to binary envelope and shaping the quantization noise out of signal band. To further suppress the noise near the in band signal that is left from noise shaping, active FIR filter structure consisting of multiple identical amplifier units is proposed. In this paper, the Channelized Active Noise Elimination (CANE) technique is employed to create an effective band-pass FIR filter by delaying, up-converting the baseband signal and then combining them in RF. A special power combining network is designed to maintain the overall PA efficiency while the noise power is suppressed. An S-band transmitter prototype with 2-channels has been built and tested. The results demonstrated that the CANE technique can achieve filtering with software reconfigured flexibility while maintaining the power efficiency.

6. **A Modified Imperialist Competitive Algorithm for Digital IIR Filter Design**

Digital infinite impulse response (IIR) filter have become the target of growing interest, because they often provide a much better performance and less computational cost than finite impulse response (FIR) filters. Since the problem of error surface of designing Digital IIR filters is generally nonlinear and multimodal, global optimization techniques are required in order to avoid local minima. In this paper, an evolutionary method based on Imperialist Competitive Algorithm (ICA) has been proposed to design Digital IIR filters. By adding a step to the standard ICA algorithm, its performance has been improved in searching solution space and convergence to the global minima. Simulation results show the efficiency of the proposed method to design Digital IIR filter**.**

7. **Robust acoustic echo cancellation in the short-time fourier transform domain using adaptive crossband filters**

This paper presents a robust acoustic echo cancellation (AEC) system in the short-time Fourier transform (STFT) domain using adaptive crossband filters. The STFT-domain AEC allows for a simpler system structure compared to the traditional frequency-domain AEC, which normally requires several applications of the discrete Fourier transform (DFT) and the inverse DFT, while the robust AEC (RAEC) allows for continuous and stable filter updates during double talk without freezing the adaptive filter. The RAEC and the STFT-domain AEC have been investigated in the past in separate studies. In this work we propose a novel algorithm that combines the advantages of both approaches for robust update of the adaptive crossband filters even during double talk. Experimental results confirm the benefit of incorporating the robustness constraint for the adaptive crossband filters and show improved performance in terms of the echo reduction and the predicted sound quality..

8. **Analysis of normal and epileptic EEG signals with filtering methods**

EEG sinyalleri, beyin yüzeyinden alınan, küçük genlikli ve düşük frekans bandına sahip durağan olmayan biyolojik işaretlerdir. Ortamda bulunan diğer elektriksel sinyaller ile kişinin kendisinden kaynaklı istemli-istemsiz hareketleri ek gürültülere sebep olabileceğinden EEG kayıtları diğer biyolojik işaretler gibi hassas bir ölçüm gerektirmektedir. Bu çalışmada, WAG/Rij farelerine ait normal/epileptik EEG sinyallerinde bulunan yüksek frekanslı gürültüler hareketli ortalama filtre ile düşük frekanslı gürültüler ise türev tabanlı filtre ile bastırılmış ve sinyallerin frekans spektrumları çıkartılmıştır. Böylece sinyallerin, yoğunlaştığı frekans bantları karşılaştırılarak hastalık teşhisinin kolaylaştırılması amaçlanmıştır.

9. **Automatic Identification and Removal of Ocular Artifacts inEEG—Improved Adaptive Predictor Filtering for PortableApplications**

Electroencephalogram (EEG) signals have a long history of use as a noninvasive approach to measure brain function. An essential component in EEG-based applications is the removal of Ocular Artifacts (OA) from the EEG signals. In this paper we propose a hybrid de-noising method combining Discrete Wavelet Transformation (DWT) and an Adaptive Predictor Filter (APF). A particularly novel feature of the proposed method is the use of the APF based on an adaptive autoregressive model for prediction of the waveform of signals in the ocular artifact zones. In our test, based on simulated data, the accuracy of noise removal in the proposed model was significantly increased when compared to existing methods including: Wavelet Packet Transform (WPT) and Independent Component Analysis (ICA), Discrete Wavelet Transform (DWT) and Adaptive Noise Cancellation (ANC). The results demonstrate that the proposed method achieved a lower mean square error and higher correlation between the original and corrected EEG. The proposed method has also been evaluated using data from calibration trials for the Online Predictive Tools for Intervention in Mental Illness (OPTIMI) project. The results of this evaluation indicate an improvement in performance in terms of the recovery of true EEG signals with EEG tracking and computational speed in the analysis. The proposed method is well suited to applications in portable environments where the constraints with respect to acceptable wearable sensor attachments usually dictate single channel devices.

10. Towards Generalizing Classification Based Speech Separation

Monaural speech separation is awell-recognized challenge. Recent studies utilize supervised classificationmethods to estimate the ideal binary mask (IBM) to address the problem. In a supervised learning framework, the issue of generalization to conditions different from those in training is very important. This paper presents methods that require only a small training corpus and can generalize to unseen conditions. The system utilizes support vector machines to learn classification cues and then employs a rethresholding technique to estimate the IBM. A distribution fittingmethod is used to generalize to unseen signal-to-noise ratio conditions and voice activity detection based adaptation isused to generalize tounseen noise conditions. Systematic evaluation and comparison show that the proposed approach produces high quality IBM estimates under unseen conditions.