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Title with Abstracts

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Abstract: Digital signal processing (DSP) plays several important roles in modern radio astronomy, such as processing data to create high-resolution radio images, isolating weak emissions from celestial sources, and reducing distortions in incoming signals. DSP can also manage beam-forming, a complex process that allows radio signals to be received from across the sky from any direction, and even multiple directions simultaneously.

Abstract: Developing audio processing tools for extracting social-audio features are just as important as conscious content for determining human behavior. Psychologists speculate these features may have evolved as a way to establish hierarchy and group cohesion because they function as a subconscious discussion about relationships, resources, risks, and rewards. In this paper, we present the design, implementation, and deployment of a wearable computing platform capable of automatically extracting and analyzing social-audio signals. Unlike conventional research that concentrates on data which have been recorded under constrained conditions, our data were recorded in completely natural and unpredictable situations. In particular, we benchmarked a set of integrated algorithms (sound speech detection and classification, sound level meter calculation, voice and nonvoice segmentation, speaker segmentation, and prediction) to obtain speech and environmental sound social-audio signals using an in-house built wearable device. In addition, we derive a novel method that incorporates the recently published audio feature extraction technique based on power normalized cepstral coefficient and gap statistics for speaker segmentation and prediction. The performance of the proposed integrated platform is robust to natural and unpredictable situations. Experiments show that the method has successfully segmented natural speech with 89.6% accuracy.
### ETPL SP - 003 Optimizing a High-Order Graphic Equalizer for Audio Processing

Abstract: A high-order graphic equalizer has the advantage that the gain in one band is highly independent of the gains in the adjacent bands. However, all practical filters have transition bands, which interact with the adjacent bands and create errors in the desired magnitude response. This letter proposes a filter optimization algorithm for a high-order graphic equalizer, which minimizes the errors in the transition bands by iteratively optimizing the orders of adjacent band filters. The optimization of the filter order affects the shape of the transition band, thus enabling the search for the optimum shape relative to the adjacent filter. The optimization is done offline, and during filtering only the gains of the band filters are altered. In an example case, the proposed method was able to meet the given peak-error limitations of ±2 dB, when the total order of the graphical equalizer was 328, whereas the non-optimized filter could not meet the requirements even when the total order was raised to 672. Optimized high-order graphical equalizers can be widely used in audio signal processing applications.

### ETPL SP - 004 Speech Processing on a Reconfigurable Analog Platform

Abstract: We describe architectures for audio classification front ends on a reconfigurable analog platform. Real-time implementation of audio processing algorithms involving discrete-time signals tend to be power-intensive. We present an alternate continuous-time system implementation of a noise-suppression algorithm on our reconfigurable chip, while detailing the design considerations. We also describe a framework that enables future implementations of other speech processing algorithms, classifier front ends, and hearing aids.
**ETPL SP - 005**  
**Reduction of Signal-Dependent Noise From Hyperspectral Images for Target Detection**

Abstract: Tensor-decomposition-based methods for reducing random noise components in hyperspectral images (HSIs), both dependent and independent from signal, are proposed. In this paper, noise is described by a parametric model that accounts for the dependence of noise variance on the signal. This model is thus suitable for the cases where photon noise is dominant compared with the electronic noise contribution. To denoise HSIs distorted by both signal-dependent (SD) and signal-independent (SI) noise, some hybrid methods, which reduce noise by two steps according to the different statistical properties of those two types of noise, are proposed in this paper. The first one, named as the PARAFACSI- PARAFACSD method, uses a multilinear algebra model, i.e., parallel factor analysis (PARAFAC) decomposition, twice to remove SI and SD noise, respectively. The second one is a combination of the well-known multiple-linear-regression-based approach termed as the HYperspectral Noise Estimation (HYNE) method and PARAFAC decomposition, which is named as the HYNE-PARAFAC method. The last one combines the multidimensional Wiener filter (MWF) method and PARAFAC decomposition and is named as the MWF-PARAFAC method. For HSIs distorted by both SD and SI noise, first, most of the SI noise is removed from the original image by PARAFAC decomposition, the HYNE method, or the MWF method based on the statistical property of SI noise; then, the residual SD components can be further reduced by PARAFAC decomposition due to its own statistical property. The performances of the proposed methods are assessed on simulated HSIs. The results on the real-world airborne HSI Hyperspectral Digital Imagery Collection Experiment (HYDICE) are also presented and analyzed. These experiments show that it is worth taking into account noise signal-dependence hypothesis for processing HYDICE data.
**Polymer Optical Fiber Termination With Use of Liquid Nitrogen**

Abstract: In noise reduction, a common approach is to use a microphone array with a beamformer that combines the individual microphone signals to extract a desired speech signal. The beamformer weights usually depend on the statistics of the noise and desired speech signals, which cannot be directly observed and must be estimated. Estimators based on the speech presence probability (SPP) seek to update the statistics estimates only when desired speech is known to be absent or present. However, they do not normally distinguish between desired and undesired speech sources. In this contribution, an algorithm is proposed to distinguish between these two types of sources using additional spatial information, by estimating a desired speech presence probability based on the combination of a multichannel SPP and a direction of arrival (DOA) based probability. The DOA-based probability is computed using DOA estimates for each time-frequency bin. The estimated statistics are then used to compute the weights of a spherical harmonic domain tradeoff beamformer, which achieves a balance between noise reduction and speech distortion. The performance evaluation demonstrates the effectiveness of the proposed approach at suppressing both background noise and spatially coherent noise. A number of audio examples and sample spectrograms are also provided.

**Reduction of Noise in Measurements of Phasor Angles by Using Two Digital Filters**

Abstract: An electric power system is monitored and controlled on the basis of its phasor information, such as amplitude, phase angle, and frequency. This phasor can be measured through discrete Fourier transform (DFT) coefficients of its positive nominal frequency. Then, the frequency estimate can be calculated from time variation of the measured phasor angle. The accuracy of the phasor-angle measurement and the frequency estimation are severely affected by noise in the power system signal and the leakage of the negative fundamental frequency in DFT. This paper proposes a DFT-based phasor-angle measurement algorithm to cope with both the noise effect and the leakage effect. In addition, a frequency estimation algorithm is developed to use the same technique in the phasor-angle measurement algorithm. These algorithms introduce two digital filters for the reduction of the noise effect. The effectiveness of the proposed algorithm in the noise effect reduction is verified through simulations.
ETPL SP - 008  SNR and Noise Variance Estimation in Polarimetric SAR Data

Abstract: Thermal noise affects polarimetric measurements; but a first order noise correction can be applied if the noise variance (or power) is known. This letter deals with the estimation of the noise variance and the signal-to-noise ratio (SNR) of the cross-polarized channels in polarimetric synthetic aperture radar (SAR) data. Cramér-Rao lower bounds (CRLB) and maximum likelihood (ML) estimators are derived. The ML noise variance estimator is unbiased and efficient, while the ML SNR estimator is biased, but an unbiased SNR estimator can be derived from the biased one. It is also shown that commonly used noise variance and SNR estimators are biased. The results are finally validated using TerraSAR-X fully polarimetric data.

ETPL SP - 009  Simultaneous Low-Pass Filtering and Total Variation Denoising

Abstract: This paper seeks to combine linear time-invariant (LTI) filtering and sparsity-based denoising in a principled way in order to effectively filter (denoise) a wider class of signals. LTI filtering is most suitable for signals restricted to a known frequency band, while sparsity-based denoising is suitable for signals admitting a sparse representation with respect to a known transform. However, some signals cannot be accurately categorized as either band-limited or sparse. This paper addresses the problem of filtering noisy data for the particular case where the underlying signal comprises a low-frequency component and a sparse or sparse-derivative component. A convex optimization approach is presented and two algorithms derived: one based on majorization-minimization (MM), and the other based on the alternating direction method of multipliers (ADMM). It is shown that a particular choice of discrete-time filter, namely zero-phase noncausal recursive filters for finite-length data formulated in terms of banded matrices, makes the algorithms effective and computationally efficient. The efficiency stems from the use of fast algorithms for solving banded systems of linear equations. The method is illustrated using data from a physiological-measurement technique (i.e., near infrared spectroscopic time series imaging) that in many cases yields data that is well-approximated as the sum of low-frequency, sparse or sparse-derivative, and noise components.
### Finite Delay Response Harmonic Filters

**Abstract:** This paper proposes a theory of harmonic filters and their mathematical models, applicable to periodically time varying (PTV) systems perturbed by a large periodic signal. The harmonic filters are based on a series of finite time delay and weighted sum operations, which allow selection or rejection of an input fundamental tone and/or its harmonics in a periodic manner. The harmonic filters are essentially array processors and can be analogous to digital FIR filters. The paper discusses optimum ways of choosing filter coefficients, time delays, and weights for the harmonic rejections and selections, equivalent to lowpass filters and highpass filters, respectively. Mathematical models for effects of hardware mismatches, weight and delay mismatches, on the filtering performance are provided and verified through ADS behavioral statistical simulations. The theory can be applicable to design quasi-sinusoidal mixers for a quality spectral purity, or harmonic carrier modulators and demodulators for high frequency applications.

### Design and Implementation of a Preprocessing Circuit for Bandpass Signals Acquisition

**Abstract:** The processing capabilities that are included into the acquisition block of the real-time digital oscilloscopes largely contribute to determine the overall performance of the instrument. Their remarkable improvement has made it possible to enhance the performance in terms of increased measurement rate, automation, and reduced measurement uncertainty related to quantization and noise. This paper presents the implementation of a preprocessing circuit for a novel acquisition mode of bandpass signals, which is characterized by an increased vertical resolution. Although the theoretical foundations were recently presented with simulative results, here, the circuital implementation of such an acquisition mode is presented. The focus is on mid or low cost digital oscilloscopes that can improve their vertical resolution at a negligible additional cost. First, a preliminary field programmable gate array implementation is considered to evaluate the achievable performance both from a theoretical point of view and throughout experimental tests. Then, a custom application specific integrated circuit implementation, in 28-nm complementary metal-oxide-semiconductor technology is analyzed. Along with the parameter optimization, the work experimentally tests the acquisition mode and evaluates the effects of nonideal characteristics such as finite word length and nonideal filtering. The increase in the effective number of bit (ENoB) is up to 2.5 bit, whereas the ENoB degradation because of word length and nonideal filtering is quantified as ~ 1.1 and 0.5 bit. The design highlights that there is substantial margin for parallel implementation that is the base to candidate the proposed solution as a remarkable option for the next generation oscilloscopes.
The Compression of Electric Signal Waveforms for Smart Grids: State of the Art and Future Trends

Abstract: In this paper, we discuss the compression of waveforms obtained from measurements of power system quantities and analyze the reasons why its importance is growing with the advent of smart grid systems. While generation and transmission networks already use a considerable number of automation and measurement devices, a large number of smart monitors and meters are to be deployed in the distribution network to allow broad observability and real-time monitoring. This situation creates new requirements concerning the communication interface, computational intelligence and the ability to process data or signals and also to share information. Therefore, a considerable increase in data exchange and in storage is likely to occur. In this context, one must achieve an efficient use of channel communication bandwidth and a reduced need of storage space for power system data. Here, we review the main compression techniques devised for electric signal waveforms providing an overview of the achievements obtained in the past decades. Additionally, we envision some smart grid scenarios emphasizing open research issues regarding compression of electric signal waveforms. We expect that this paper will contribute to motivate joint research efforts between electrical power system and signal processing communities in the area of signal waveform compression.

A novel multichannel audio signal compression method based on tensor representation and decomposition.

Abstract: Multichannel audio signal is more difficult to be compressed than mono and stereo ones. A novel multichannel audio signal compression method based on tensor representation and decomposition is proposed in this paper. The multichannel audio is represented with 3-order tensor space and is decomposed into core tensor with three factor matrices in the way of channel, time and frequency. Only the truncated core tensor is transmitted which will be multiplied by the pre-trained factor matrices to reconstruct the original tensor space. Objective and subjective experiments have been done to show a very noticeable compression capability with an acceptable output quality. The novelty of the proposed compression method is that it enables both high compression capability and backward compatibility with limited signal distortion to the hearing.
### ETPL SP - 014 Greedy Adaptive Linear Compression in Signal-Plus-Noise Models

Abstract: In this paper, we examine adaptive compression policies, when the sequence of vector-valued measurements to be compressed is noisy and the compressed variables are themselves noisy. The optimization criterion is information gain. In the case of sequential scalar compressions, the unit-norm compression vectors that greedily maximize per-stage information gain are eigenvectors of an a priori error covariance matrix, and the greedy policy selects them according to eigenvalues of a posterior covariance matrix. These eigenvalues depend on all previous compressions and are computed recursively. A water-filling solution is given for the optimum compression policy that maximizes net information gain, under a constraint on the average norm of compression vectors. We provide sufficient conditions under which the greedy policy for maximizing stepwise information gain actually is optimal in the sense of maximizing the net information gain. In the case of scalar compressions, our examples and simulation results illustrate that the greedy policy can be quite close to optimal when the noise sequences are white.

### ETPL SP - 015 Orthogonal Space Projection (OSP) Processing for Adaptive Interference Cancellation

Abstract: An Orthogonal Space Projection (OSP) processing methodology is introduced that achieves interference cancellation within a single time compression interval, e.g. pulse interval for fast time or dwell interval for slow time. The OSP technique can operate in pre-compression or post-compression spaces where covariance matrix development and compression are performed simultaneously. OSP projects the signal and interference into a space that is matched to the signals of interest and into a space that is mismatched to the signals of interest similar to a blocking matrix in conventional adaptive processing methods. The output of the OSP process is a two-dimensional image space that combines signal compression and interference cancellation. A pre-compression OSP technique is applied to array processing for jammer suppression and a two-dimensional array example is used to assess SINR performance.
**Abstract:** We present a novel approach to surface EMG data characterization by using time-varying multicomponent signal modeling. An EMG signal is described as a set of stationary non-harmonically related sinusoids (signal components) whose time-varying bandwidth is modeled by polynomials. The polynomial coefficients, estimated from a set of linear equations, capture the relationship between the instantaneous frequency and amplitude for individual signal components. It is proposed that such a compact EMG signal modeling may be a good candidate for a number of applications in surface electromyography: compression, muscle activity detection and low-bias conduction velocity, to name a few.