FORTY-FOURTH ASILOMAR CONFERENCE ON SIGNALS, SYSTEMS & COMPUTERS

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Welcome from the General Chairperson

Prof. Linda S. DeBrunner, Florida State University

It is my great pleasure to welcome you to the Forty-Fourth Asilomar Conference on Signals, Systems, and Computers. This conference provides a special opportunity for those of us who return year after year—to refresh our spirits and reinvigorate our research. I hope that those of you attending for the first time will find the conference as rewarding as I do. This conference provides an opportunity to share ideas with the top researchers in our field in a relaxed and friendly atmosphere. Be sure to take the opportunity to meet someone new at the family-style meals, and don't forget to enjoy a walk on the beach.

For the Sydney Parker Memorial Lecture, we are very fortunate to have a keynote address by Dr. Ronald W. Shafer, HP Fellow in the Media Communication and Networking Laboratory at Hewlett-Packard Laboratories. His talk, "A Celebration of DSP Technologies," will combine a retrospective look at the development of the discipline with a peek into the future. His talk will provide a unique framework to view the contributions of the conference this year.

The Asilomar Conference provides a supportive environment for students to present their research. This year we had 91 submissions to the student paper contest, chaired by Xinmiao Zhang. On Sunday afternoon before the Welcome Reception, the 9 finalists will present their posters to a panel of judges. I hope you have a chance to view their posters or hear their presentations during the sessions later in the week.

The success of this meeting is due to Miloš Doroslovački from The George Washington University. I want to thank him for making my job so enjoyable. He recruited outstanding technical area chairs, who then recruited outstanding session chairs. They all worked hard to create a superb technical program of 454 papers (including about 200 invited papers). I would like to thank the Technical Program Committee: Erik G. Larsson, Robert W. Heath, Jr., Ananthram Swami, Petar M. Djurić, Antonia Papandreou-Suppappola, Murray H. Loew, Miloš D. Ercegovac, David V. Anderson, and James A. Ritcey. I also want to thank all the session chairs and participants for making this another great Asilomar Conference.

Special thanks go to Sue Netzorg, Monique Fargues, Mike Matthews, Frank Kragh and Murali Tummala who perform the tasks that make this conference happen. Year after year they provide countless hours of service in arranging the venue and meals, publishing the proceedings, providing publicity, reviewing contracts and signing checks. I would like to personally thank each of them for their help and support.

I hope that you enjoy everything that Asilomar has to offer!

Linda S. DeBrunner, Florida State University, June 2010

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2010 Asilomar Conference Session Schedule

Sunday Afternoon, November 7, 2010

2:00 - 7:00 PM	Registration — Main Lodge
4:30 - 6:30 PM	Student Paper Contest — Merrill Hall
7:00 - 9:00 PM	Welcoming Dessert Reception — Merrill Hall

Monday Morning, November 8, 2010

7:30 - 9:00 AM	Breakfast – Crocker Dining Hall
8:00 AM - 6:00 PM	Registration
8:15 - 9:45 AM	MA1a — Conference Welcome and Plenary Session
9:45 - 10:15 AM	Coffee Social

10:15 AM - 12:00 PM MORNING SESSIONS

MA1b	Tensors Methods in Signal Processing
MA2b	MIMO Interference Networks
MA3b	Security in Wireless Networks
MA4b	New Trends in Sequential System Identification
MA5b	Biomotivated Recognition and Detection
MA6b	Computer Arithmetic I
MA7b	Biological Models of Speech Perception and Their Applications in Automatic Speech Processing
MA8b1	Communication Systems I (Poster)
MA8b2	Selected Topics in Image Processing (Poster)
MA8b3	Applications of Compressive Sensing (Poster)

12:00 - 1:00 PM Lunch – Crocker Dining Hall

Monday Afternoon, November 8, 2010

1:30 - 5:10 PM AFTERNOON SESSIONS

MP1a	Interference Channels
MP1b	Trends for Future Wireless Systems
MP2a	MIMO Secrecy
MP2b	MIMO Relays
MP3a	New Trends in Information Theory and Networks
MP3b	Learning and Optimization in Dynamic Networks
MP4a	Biomedical Image Analysis
MP4b	Advances in Adaptive Algorithms
MP5	Statistical Signal Processing for Complex Systems
MP6	Communication Processors and Accelerators
MP7a	Video Compression
MP7b	Advances in Keyword Spotting
MP8a1	Communication Systems II (Poster)
MP8a2	Speech Enhancement (Poster)
MP8a3	Selected Topics in Speech and Audio (Poster)
MP8a4	Adaptive Signal Processing in Communications (Poster)
MP8a5	Array-based Estimation (Poster)

Monday Evening, November 8, 2010

6:00 - 9:30 PM Conference Cocktail/Social — Merrill Hall
The Cocktail/Social takes the place of Monday's dinner. No charge for conference attendees or their guests.

2010 Asilomar Conference Session Schedule

(continued)

Tuesday Morning, November 9, 2010

7:30 - 9:00 AM	Breakfast — Crocker Dining Hall
8:00 AM - 5:00 PM	Registration

8:15 - 12:00 PM	MORNING SESSIONS
TA1a	Network Error Correction and Physical Layer Security
TA1b	Coding
TA2a	Signal Processing for Communications Receivers
TA2b	Communications Under Doppler Spread
TA3a	Recursive Reconstruction of Sparse Sequences
TA3b	Self-Organizing Networks: Architectures, Protocols and Algorithms
TA4a	Shape and Time in Biomedical Images
TA4b	Mathematical Methods for Biomedical Signals and Images
TA5	Compressive Sensing
TA6a	Reconfigurable Architectures, Algorithms and Applications
TA6b	Array Processing and Beamforming
TA7	Image and Video Enhancement
TA8a1	Cooperative and Cognitive Transmission in Multi-Antenna Networks I (Poster)
TA8a2	Cognitive Networking (Poster)
TA8a3	Adaptive Signal Processing: Theory and Applications (Poster)
TA8b1	Cooperative and Cognitive Transmission in Multi-Antenna Networks II (Poster)
TA8b2	Architectures, Implementations, and Tools I (Poster)
TA8b3	Architectures, Implementations, and Tools II (Poster)

12:00 - 1:00 PM	Lunch – Crocker Dining Hall

Tuesday Afternoon, November 9, 2010

1:30 - 5:10 PM	AFTERNOON SESSIONS
TP1a	Advances in Multihop and Distributed Wireless Transmission
TP1b	Wireless Communications
TP2a	MIMO Underwater Acoustic Communications
TP2b	MIMO for Ad Hoc Networks
TP3a	Non-Stationary Processing of Environments
TP3b	Network Information Theory
TP4a	Modeling for Biomedical Imaging
TP4b	Adaptive Filters - Theory and Applications
TP5a	Statistical Signal Processing for Neural Signals
TP5b	Integrated Multimodal Sensing
TP6a	Computer Arithmetic II
TP6b	Computer Arithmetic III
TP7a	Microphone Array Processing for Speech Applications I
TP7b	Microphone Array Processing for Speech Applications II
TP8a1	Low Complexity Implementation and Receiver Issues (Poster)
TP8a2	Detection & Estimation in Networks (Poster)
TP8a3	Techniques in Networking and Communications (Poster)
TP8b1	Scheduling, Relaying and Routing (Poster)
TP8b2	Statistical and Adaptive Signal Processing (Poster)
TP8b3	Biomedical Signals and Images (Poster)

Tuesday Evening, November 9, 2010

No conference planned event, please enjoy the Monterey Peninsula.

2010 Asilomar Conference Session Schedule

(continued)

Wednesday Morning, November 10, 2010

7:30 - 9:00 AM	Breakfast — Crocker Dining Hall
8:00 AM - 12:00 PM	Registration — Copyright Forms must be turned in before the registration closes at 12:00 noon.

8:15 AM - 12:00 PM MORNING SESSIONS

WA1a	Cooperative Communications
WA1b	Communication Theory
WA2a	Interference Management I
WA2b	Interference Management II
WA3a	Sensor Networks
WA3b	Multiuser Beamforming and Interference Channels
WA4	Advances on Adaptive Filtering and Applications
WA5	Statistical Signal Processing
WA6a	Estimation and Detection
WA6b	SOC Architectures and Applications
WA7a	Sparse Representations in Image Processing
WA7b	MIMO Radar

12:00 - 1:00 PM Lunch — Meal tickets may be purchased at registration desk. This meal is not included in the registration.

Student Paper Contest

Merrill Hall - Sunday, November 7, 2010, 4:30 - 6:30 PM

"Outage Probability of MISO Broadcast Systems with Noisy Channel Side Information"
Alon Shalev Housfater, Teng Joon Lim, University of Toronto

"Distributed Learning under Imperfect Sensing in Cognitive Radio Networks"
Keqin Liu, Qing Zhao, University of California, Davis; Bhaskar Krishnamachari, USC

"Biologically Inspired Coupled Antenna Array for Direction of Arrival Estimation"
Murat Akcakaya, Washington University in St. Louis; Carlos H. Muravchik, Universidad Nacional de La Plata; Arye Nehorai, Washington University in St. Louis

"Weighted Sum-Rate Maximization for a Set of Interfering Links via Branch and Bound"
Chathuranga Weeraddana, Marian Codreanu, Matti Latva-aho, University of Oulu; Anthony Ephremides, University of Maryland

"A Low Energy High Speed Reed-Solomon Decoder Using Decomposed Inversionless Berlekamp-Massey Algorithm"
Hazem A.Ahmed, Hamed Salah, Tallal ElShabrawy, German University in Cairo; Hossam A. H. Fahmy, Cairo University

"ρ-Domain Rate Control for JPEG XR"
Duncan Chan, Jie Liang, Simon Fraser University; Chengjie Tu, Microsoft Corp.

"Achievable Rates in Two-user Interference Channels with Finite Inputs and (Very) Strong Interference"
Frederic Knabe, Aydin Sezgin, Ulm University

"Distributed Signature Learning and Calibration for Large-Scale Sensor Networks"
Naveen Ramakrishnan, Emre Ertin, Randolph Moses, The Ohio State University

"The Role of Channel Distribution Information in the Cross-Layer Design of Opportunistic Scheduler for MIMO Networks"
Sheu-Sheu Tan, University of California, San Diego; Adam Anderson, University of South Florida; James Zeidler, University of California, San Diego

Coffee breaks will be at 9:55 AM and 3:10 PM. (Except Monday morning when refreshments will be served outside Merrill Hall from 9:45–10:15 AM)

Tuesday, November 8, 2010

CONFERENCE OPENING AND PLENARY SESSION 8:15 – 9:45 AM

1. Welcome from the General Chairperson:

Prof. Linda DeBrunner
Florida State University

2. Session MA1a Distinguished Lecture for the 2010 Asilomar Conference

A Celebration of DSP Technology

Dr. Ronald W. Schafer
Multimedia Communication and Networking Lab
Hewlett-Packard Laboratories
Palo Alto, CA 94304

Abstract
DSP is an indispensable technology with widespread impact in many areas of application; however, it has take 60 years or more to get to where we are today. Thus, it may be interesting and worthwhile to take a look at how the DSP technology domain originated and evolved. In this talk, I will look back at some of what I consider to be the most important milestones and the people behind them, examine some of the key interactions with other technologies, consider the importance of unfettered application-centric research, and comment on the importance of education in the evolution of DSP. The goal of this analysis is to provide a platform from which to admire and celebrate the past progress and make guesses about what the future might hold for the field of DSP.

Biography
Ronald W. Schafer received BSEE (1961) and MSEE (1962) degrees from the University of Nebraska and a Ph.D. (1968) degree from MIT. From 1968 to 1974 he was a member of the Acoustics Research Department, Bell Laboratories, Murray Hill, NJ, where he contributed to some of the earliest research on digital signal processing. In 1974 he joined Georgia Tech as John and Marilu McCarty Professor of Electrical and Computer Engineering. Over a thirty-year academic career, he introduced literally thousands of students to the field of digital signal processing and supervised graduate student research in speech processing, image processing, biomedical signal processing, and communication signal processing. He played a major role in establishing the Center for Signal and Image Processing at Georgia Tech as a major force in DSP education and research, and in 1982 he co-founded Atlanta Signal Processors, Inc., one of the first companies to provide design tools for DSP systems.

Dr. Schafer retired from Georgia Tech as Professor Emeritus in 2004. Now he is a HP Fellow in the Multimedia Communication and Networking Laboratory at Hewlett-Packard Laboratories in Palo Alto, CA, where his research focuses on acoustic signal processing and immersive communications.

Dr. Schafer is a Fellow of the IEEE and the Acoustical Society of America, and he is a member of the National Academy of Engineering. He has co-authored numerous widely used textbooks including Digital Signal Processing (1975), Digital Processing of Speech Signals (1978), Signal Processing First (2003), Discrete-Time Signal Processing (2009), and Theory and Application of Digital Speech Processing (2010). He has received numerous awards for teaching and research including the 1985 Distinguished Professor Award from Georgia Tech, the 1980 IEEE Emanuel R Piori Award, the 1992 IEEE James H. Mulligan, Jr. Education Medal, and he received the 2010 IEEE Jack S. Kilby Medal.

Program of 2010
Asilomar Conference
on
Signals, Systems, and Computers

Technical Program Chairman
Prof. Miloš Doroslovački
The George Washington University

Track 1 – A. Communications Systems

Session: MAb1 – Tensors Methods in Signal Processing

Chair: ***Martin Haardt,*** *Technical University Ilmenau*

MA1b-1 10:15 AM

Overview of Recent Advances in Numerical Tensor Algebra

Göran Bergqvist, Erik G. Larsson, Linköping University

We present a survey of some recent developments for decompositions of multi-way arrays or tensors, with special emphasis on results relevant for applications and modeling in signal processing. A central problem is how to find low-rank approximations of tensors, and we will describe new results, including numerical methods, algorithms and theory, for the higher order singular value decomposition and the parallel factors expansion (canonical decomposition).

MA1b-2 10:40 AM

Blind Estimation of SIMO Channels Using A Tensor-Based Subspace Method

Bin Song, Florian Roemer, Martin Haardt, Ilmenau University of Technology

We introduce a tensor-based subspace method for solving the blind channel estimation problem in a single-input multiple-output (SIMO) system. Since the measurement data is multidimensional, previously proposed blind channel estimation methods require stacking the dimensions into one highly structured matrix and estimate the signal subspace via a singular value decomposition (SVD) of the correlation matrix of the measurement data. In contrast to this, we define a 3-way measurement tensor of the received signals and obtain the signal subspace via a Higher-Order SVD (HOSVD). This allows us to exploit the structure inherent in the measurement data and leads to better estimates of the signal subspace. Numerical simulations demonstrate that the proposed method outperforms previous methods in terms of the channel estimation accuracy.

MA1b-3 11:05 AM

New Simultaneous (Generalized) Schur Decomposition Methods for the Computation of the Canonical Polyadic Decomposition

Mikael Sorensen, University of Nice; Lieven De Lathauwer, K.U. Leuven

In signal processing several problems have been formulated as Simultaneous Generalized Schur Decomposition (SGSD) problems. Applications are found in blind source separation and multidimensional harmonic retrieval. Furthermore, SGSD methods for computing a third-order Canonical Polyadic (CP) decomposition have been proposed. The SGSD method mainly requires that two of the matrix factors of the CP decomposition have full column rank. We first propose a new version of the SGSD method for computing a third-order CP decomposition. The proposed method mainly differs from the existing methods in the way the triangular matrices are computed. Second, we explain how to extend this method to tensors of order higher than three. Third, we also explain how the proposed method can be used for tensors with partial (Hermitian) symmetry.

MA1b-4 11:30 AM

A k-dimensional Subspace-based Tensor Factorization Approach for Underdetermined Blind Identification

Bahador Makki Abadi, Saeid Sanei, Dave Marshall, Cardiff University

In linear blind source separation, if the number of sources is more than the number of mixtures, for estimation of the mixing matrix and consequently the sources, often a single dominant component sparse component analysis (SCA) is used. In typical applications this is not feasible since more than one source may be active at each instant. In a multiple dominant components SCA, the segments with k active sources are identified. Here, a k-subspaces approach is developed using tensor factorization. The identified segments are used in the design of a tensor. The mixing matrix is then estimated and used for extraction of the sources.

Session: MAb2 – MIMO Interference Networks

Chair: ***Syed Jafar,*** *University of California, Irvine*

MA2b-1 10:15 AM

On Relay-Interference Networks with Quantized Feedback

Erdem Koyuncu, Hamid Jafarkhani, University of California, Irvine

We study quantized beamforming in wireless relay-interference networks with multiple transmitter-receiver pairs. For given transmitter rate requirements, we design structured distributed quantizers specifically to optimize the symbol error rate performance. We show that our quantizers achieve both maximal diversity and very high array gain using arbitrarily low feedback rates per receiver. Simulations are also provided, confirming our analytical results. We observe that our quantizers guarantee an equal diversity gain for each transmitter-receiver pair.

MA2b-2 10:40 AM

Connecting Interference Alignment and Distributed Storage Through Rank Minimization

Dimitris Papailiopoulos, Alexandros Dimakis, University of Southern California

We show that the maximization of the sum degrees-of-freedom (DoF) for the static flat-fading multiple-input multiple-output (MIMO) interference channel is equivalent to a rank constrained rank minimization (RCRM) problem when the signal spaces are required to span a fixed number of dimensions. The rank minimization corresponds to maximizing interference alignment (IA) such that interference spaces span the lowest dimensional subspace possible. The rank constraints account for the useful signal spaces spanning all spatial dimensions available. Inspired by recent results in low-rank matrix completion theory, we show that the convex envelope of the sum of ranks of the interference matrices is the sum of their corresponding nuclear norms. Moreover, we relax the rank constraints to asymptotically equivalent convex ones. We further show that for the multi-cell interference channel a similar RCRM problem with extra affine constraints can be formulated that is equivalent to maximizing the spatial DoF. Using such formulation and IA arguments, we establish a connection between the spatial DoF maximization for the multi-cell scenario and the minimization of the total bandwidth communicated to exactly repair the data nodes of a given distributed storage MDS coded system. We provide a mapping which suggests that achieving certain DoF for the first problem provides a bandwidth bound for the second and under conditions vice versa.

MA2b-3 11:05 AM

Real Interference Alignment

Abolfazl Motahari, Shahab Oveisgharan, Mohammad Ali Maddah-Ali, Amir Khandani, University of Waterloo

In this paper, we show that the total Degrees-Of-Freedoms (DOF) of the K-user Gaussian Interference Channel (GIC) can be achieved by incorporating a new alignment technique known as “Real Interference Alignment” (RIA). This technique compared to its ancestor, namely “Vector Interference Alignment”, exploits a single real line and relies on the properties of real numbers to align the multi-user interference. The RIA relies on a new scheme in which several data streams having fractional multiplexing gains are sent by transmitters and interfering streams are aligned at the receivers. The proposed scheme is backed up by a recent result in the field of Diophantine approximation, which states that the convergence part of the Khintchine-Groshev theorem holds for points on non-degenerate manifolds.

MA2b-4 11:30 AM

On the Capacity of a Class of Degraded MIMO Z Interference Channels with Degraded Message Sets

Fabio Fernandes, Sriram Vishwanath, University of Texas at Austin

In this paper, a MIMO Z Interference Channel (IFC) with degraded message sets is considered. Such a degraded MIMO Z IFC forms an outer bound on the capacity of the of MIMO cognitive radio channel. In this paper, we use Fisher-information based inequalities to show that, for certain classes, Gaussian inputs can be shown to be optimal for this channel.

Track 3 – C. Networks

Session: MAb3 – Security in Wireless Networks

Co-Chairs: ***Dennis Goeckel,*** *University of Massachusetts, Amherst* **and Don Towsley,** *University of Massachusetts, Amherst*

MA3b-1 10:15 AM

From Uncertainty to Secrecy: A Dynamic Approach

Sheng Xiao, Weibo Gong, Donald Towsley, University of Massachusetts

Uncertainty is the pseudonym for many unfavorable effects in both wired and wireless communications. When uncertainty presents, a receiver suffers from information loss and the communication efficiency is limited. Not only legitimate users, the adversary also faces inevitable information loss. This information loss can be utilized for security purposes. In this paper, we introduce the framework of utilizing adversary's information loss as a security measure. Moreover, we propose a set of light-weight algorithms to generate a series of hash values, namely dynamic secrets, from communication traffic and then apply dynamic secrets to secure the communications.

MA3b-2 10:40 AM

Secrecy Coverage

Amites Sarkar, Western Washington University; Martin Haenggi, University of Notre Dame

Recently we have proposed a geometric model, the so-called secrecy graph, to quantify the impact of the presence of eavesdroppers on the connectivity of wireless networks. Here we are using a similar model to assess their impact on the coverage of a network of base stations. The question we address is the following: Let base stations and eavesdroppers be distributed as stationary Poisson point processes in a square of area n. If the coverage of each base station is limited by the distance to the nearest eavesdropper, what is the maximum density of eavesdroppers that can be accommodated while still achieving full coverage, asymptotically as n goes to infinity?

MA3b-3 11:05 AM

Control of Wireless Networks with Secrecy

C. Emre Koksal, Ohio State University; Ozgur Ercetin, Yunus Sarikaya, Sabanci University

We consider the problem of cross-layer resource allocation in time-varying cellular and multihop wireless networks, and incorporate information theoretic secrecy as a Quality of Service constraint. Specifically, each node in the network injects two types of traffic, secure and unsecure, at rates chosen in order to maximize a global utility function, subject to network stability and secrecy constraints. The secrecy constraint enforces an arbitrarily low mutual information leakage from the source to every node in the network, except for the sink node. We show that the secrecy constraint leads to solutions that are significantly different from their counterparts without secrecy. We focus on a set of topologies and develop an end-to-end secure coding scheme along with the associated efficient routing and scheduling schemes to achieve a high performance. While we assume the nodes have full CSI of their neighbors, we generalize our results to the case with limited CSI.

MA3b-4 11:30 AM

Embedding Covert Information Flow

Stefano Marano, Vincenzo Matta, Unversity of Salerno; Lang Tong, Cornell University

We consider the problem of embedding a information flow in multi-hop wireless transmissions so that the transmission timing does not reveal the presence of such a flow. We establish a connection between the problem of embedding covert information flow in renewal processes and the Riemann-Hilbert boundary value problem, which allows us to obtain accurate evaluation of the maximum achievable rate of embedding.

Track 4 – D. Adaptive Systems and Processing

Session: MAb4 – New Trends in Sequential System Identification

Chair: ***Cédric Richard,*** *Université de Nice Sophia-Antipolis*

MA4b-1 10:15 AM

Adaptive Systems of Particle Filters

Petar Djuric, Mónica Bugallo, Stony Brook University

We study systems of particle filters that track targets based on data acquired from sensor networks. The number of particle filters in the systems varies in that more particle filters may be added to the systems, some may be removed, and others may be merged or split with time. The decisions for changing the number of filters in the systems depend on the estimated state of the targets that are being tracked. We compare the performance of these filters with ones that have one particle filter, which tracks states with possibly changing dimensionalities.

MA4b-2 10:40 AM

PARTICLE FLOW FOR NONLINEAR FILTERS: SEVENTEEN DUBIOUS SOLUTIONS TO A FIRST ORDER LINEAR UNDERDETERMINED PDE

Fred Daum, Jim Huang, Raytheon Company

We have invented a new theory of exact particle flow for nonlinear filters. This generalizes our theory of particle flow that is already many orders of magnitude faster than standard particle filters and which is several orders of magnitude more accurate than the extended Kalman filter for difficult nonlinear problems. The new theory generalizes our recent log-homotopy particle flow filters in three ways: (1) the particle flow corresponds to the exact flow of the conditional probability density corresponding to Bayes'rule; (2) roughly speaking, the old theory was based on incompressible particle flow (like subsonic flight in air), whereas the new theory allows compressible flow (like supersonic flight in air); (3) the old theory suffers from obstruction of particle flow as well as singularities in the equations for flow, whereas the new theory has no obstructions and no singularities.

MA4b-3 11:05 AM

Non-linear Adaptive Filtering with Kernel Functions: An Overview

Weifeng Liu, University of Florida; Cédric Richard, Université de Nice Sophia-Antipolis; José Princìpe, University of Florida; Simon Haykin, McMaster University

Dynamic system modeling has played a crucial role in the development of techniques for stationary and non-stationary signal processing. Most existing approaches have been built on the following three pillars due to their inherent simplicity from conceptual and implementational points of view: the linear model, the mean-squared error criterion, and the adaptive least-square learning algorithm. However, there are many practical situations, e.g., in communications and biomedical engineering, where the non-linear adaptive processing of signals is needed. Kernel-based algorithms have been a topic of considerable interest in the machine learning community over the last ten years. Their attractiveness resides in their elegant treatment of nonlinear problems. This paper presents a comprehensive introduction to non-linear adaptive filtering drawn on the theory of reproducing kernel Hilbert spaces. We use this framework to derive the kernel least-mean-square algorithm. We also introduce briefly the kernel affine projection algorithms and the kernel recursive least-squares algorithm. Next, we address the main bottleneck of kernel adaptive filters, i.e., their growing structure. Finally, we demonstrate through experiments on real and synthetic data the effectiveness of the proposed methods compared to conventional approaches.

MA4b-4 11:30 AM

On Attributes of the CKF and its Relationship to the UKF

Simon Haykin, McMaster University

The cubature Kalman filter is a new nonlinear sequential state estimator, the derivation of which builds on the cubature rule in mathematics, hence the name of the filter. As such, the CKF has a rigorous mathematical basis. Most importantly, the CKF is the best known approximation to the optimal Bayesian filter under the Gaussian assumption. It is “best” in the sense that it preserves second-order information about the hidden state, given the observables (measurements). Moreover, the CKF has some other unique properties that make it the “method of choice” for nonlinear sequential state estimation. The superior performance of the CKF over other estimators, namely, the extended Kalman filter (EKF) and unscented Kalman filter (UKF), has been demonstrated experimentally (through computer simulations) under challenging environmental conditions.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: MAb5 – Biomotivated Recognition and Detection

Chair: ***Visar Berisha,*** *Raytheon Missile Systems*

MA5b-1 10:15 AM

Evaluating Brain Software Simulations using Common Test Suite

Richard Hammet, David V. Anderson, Georgia Institute of Technology

There are many software packages are now available that simulate the function of portions of the human brain. These vary widely in the method of simulation, including which features of the brain they attempt to simulate and what type of algorithms and data structures they use to represent these features and interact with the external world. In this paper, we evaluate several of these packages for their fidelity to known neuroscience and neuropsychology. In addition, we have also tested each of these packages with a common suite of engineering problems, presenting the problems as identically as possible to each software package. We have evaluated each package for the accuracy and usefulness of its output results.

MA5b-2 10:40 AM

Making Decisions About Unseen Data: Semi-Supervised Learning at Different Levels of Specificity

Visar Berisha, Raytheon Company; Ailar Javadi, Alexander Gray, David V. Anderson, Richard Hammet, Georgia Institute of Technology

An important, yet largely under-explored, problem in pattern recognition concerns learning from data labeled at varying levels of specificity. The majority of existing machine learning methods are based on the inductive learning paradigm, where a labeled training set (one label per training example) trains a classifier. This is markedly different from the human learning experience, where any one object can take multiple labels (i.e. a dog is a dog, but it is also an animal and a living object). As a result, we propose an algorithm that considers the "classification" problem as a special case of the more general "categorization" problem. In this talk, we present a semi-supervised algorithm that can incorporate data with an (unknown) taxonomy of labels to learn a categorical representation. We show preliminary results on three data sets and compare the results of one of the data sets to human subject experiments.

MA5b-3 11:05 AM

High Resolution Radar Analysis of Human Gait

Gerald Benitz, Shourov Chatterji, Daniel Gilbert, Paul Monticciolo, Rowland O'Flaherty, Mikael Yamaguchi, Aimee D'Onofrio, MIT Lincoln Laboratory

ABSTRACT

MA5b-4 11:30 AM

Using Machines to Improve Human Saliency Detection

Nikhil Rao, Tyler Karrels, Robert Nowak, Tim Rogers, University of Wisconsin-Madison

Given an image , humans are adept at identifying informative regions. But given an entire corpus of images, it is a slow and often tedious task to analyze every image in the corpus, and identify the salient parts. A machine on the other hand, can sift through a large amount of data quickly , but methods of identifying salient regions using machines are unreliable , due to the subjectivity of human perception involved in the task. In this paper, we develop various methods to rank the images according to content and investigate how human-machine teams perform the task of searching through a large database of images and identifying salient regions.

Session: MAb6 – Computer Arithmetic I

Chair: ***M. Schulte,*** *AMD Research and Advanced Development Labs*

MA6b-1 10:15 AM

Arithmetic Techniques Employed in the Next-Generation AMD FPU Core

Debjit Das Sarma, Advanced Micro Devices; David Oliver, Veloce Technologies; Alexandru Fit-Florea, NVidia; Scott Hilker, Kevin Hurd, Kelvin Goveas, Jay Fleischman, Mark Gibson, Michael Estlick, Advanced Micro Devices

AMD's next-generation core is a power-efficient, cluster-based, multithreaded execution engine. It is AMD's flagship processor designed to enable exceptional performance/watt for a wide range of client and server applications. AMD designed the FPU in its next-generation core to deliver industry-leading performance on HPC, multimedia and gaming applications. The primary means of achieving these performance levels are the four-wide, two-way-multithreaded and fully out-of-order FPU and the presence of two 128-bit fused multiply-accumulate (FMAC) units supported by a 128-bit high-bandwidth load-store subsystem. The FPU supports the 64-bit AMD64 ISA in addition to bringing a number of advanced extensions (SSE4.1, SSE4.2, AVX, and XOP including four-operand FMAC, integer multiply-accumulate (IMAC), and permute instructions) to the AMD roadmap. The major execution units are the two 128-bit FMAC units, two 128-bit packed-integer units, one 128-bit IMAC unit and one 128-bit permute unit. The FMAC unit implements fully pipelined fused multiply-accumulate operations with one rounding. The FMAC unit also executes fully pipelined pure FADD and FMUL operations. FMAC-based divide and square-root operations are implemented with a state machine in the FMAC unit, and are not fully pipelined. A variety of micro-architectural and arithmetic techniques are employed in the FPU to achieve high performance and power efficiency. Some of these techniques include FMAC unrounded bypass in mixed precisions; sharing the double-precision FMAC hardware to execute pair-wise single-precision FMAC; using power-efficient methods for executing FADD and FMUL in the FMAC unit; implementing FMAC-based divide, square root, and transcendental instructions; novel FMAC-based rounding techniques for divide and square root; fast conversion algorithms between floating point and integers; fast packed-integer multiply-accumulate algorithms; and hardware-assisted fast encryption support.

MA6b-2 10:40 AM

Design and FPGA Implementation of Radix-10 Combined Division/Square Root Algorithm with Limited Precision Primitives

Miloš D. Ercegovac, University of California, Los Angeles; Robert McIlhenny, California State University Northridge

We present a radix-10 fixed-point digit-recurrence algorithm for combined division and square root operations using limited-precision multipliers, adders, and table-lookups. The square-root algorithm, except in the initialization steps, uses the digit-recurrence algorithm for division with limited-precision primitives, facilitating implementation of the combined scheme. We discuss the proposed combined algorithm, a design, and its FPGA implementation on a Xilinx FPGA device. We present the cost and delay characteristics for precisions of 7 (single-precision), 8, 14 (double-precision), 16, 24, and 32 decimal digits. The proposed scheme uses short (2-3 digit-wide) operators which leads to compact modules, and may have an advantage at the layout level as well as in power optimization. The proposed approach is general and can be adapted to other higher radix combined divide/square root implementations.

MA6b-3 11:05 AM

Assessment of Butterfly Network VLSI Shifter Circuit

Neil Burgess, University of Bristol

Recently, a new proposal for implementing shifters in VLSI using a butterfly network has been published. This paper assesses the claims made for this shifter when compared with a conventional shifter constructed as a number of rows of multiplexers each with a common control signal. Comparisons of area and delay are made both by using the Logical Effort delay model and by simulation of a manully placed implementation in a 65nm CMOS VLSI desigh flow.

MA6b-4 11:30 AM

An Optimized Recursive High Radix Divide Unit with Multipartite Memory Lookup

James Stine, Amey Phadke, Surpriya Tike, Justin Remington, Oklahoma State University

The use of division is an important operation for scientic computing. Although early computers utilized division units that operated serially, the more recent trend is to move toward functional units that can operate with higher amounts of algorithmic complexity. For example, some modern microprocessors utilize multiplicative-divide units that employ Goldschmidt's algorithm for division, so that every iteration converges quadtratically towards its final result. Unfortunately, multiplicative divide

algorithms consume larger amounts of area for single-core computing architectures over recursive-based dividers due to the size of the multiplier required for their operation. In today's high-performance computing environment, computer architectures are migrating towards multiple smaller computing cores per chip in hopes that the parallelization of algorithms can offset the design challenges for faster cycle times associated with nanometer design rules. Although algorithms that converge linearly, such as recursive division units, do not compute eciently as multiplicative-divide algorithms, they can still be designed such that the radix increase allows more bits to be retired per iteration. In addition, if scaling is used to eciently oset the quotient selection table size, certain recursive division algorithms can be more ecient per unit area than multiplicative division algorithms. This paper discusses a recursive radix 512 divider that is optimized with multipartite tables within the scaling unit to reduce the overall memory requirements. Results utilizing a 65nm low-power, high-performance technology indicate an average 93% reduction in memory size along with a sub-1:0ns cycle time. Layout-extracted comparisons are presented for 32-bit and 64-bit operands for both Goldschmidt and recursive-based divide algorithms designed with 65nm and 45nm minimum feature sizes.

Track 8 – H. Speech, Image and Video Processing

Session: MAb7 – Biological Models of Speech Perception and Their Applications in Automatic Speech Processing

Chair: ***Nima Mesgarani,*** *Johns Hopkins University*

MA7b-1 — 10:15 AM

Frequency Domain Perceptual Linear Prediction (FDPLP)

Hynek Hermansky, Sriram Ganapathy, Samuel Thomas, The Johns Hopkins University

Psychophysics and physiology of hearing provides significant evidence that human hearing is equipped with means for evaluating spectral dynamics of sounds within a span of several hundreds of ms. Following this lead, we have been pursuing for a number of years information extraction from series of short-term spectra or from spectra computed in longer temporal windows. More recently we apply techniques for autoregressive modeling of Hilbert envelopes in frequency sub-bands that allow for bypassing the short-term analysis entirely. We show that this technique allows for convenient and straightforward modeling of spectral dynamics in long segments of the signal, providing new means for modeling of speech spectral dynamics in coding an recognition of acoustic signals and opening directions for dealing with harmful effects of linear distortions and reverberations.

MA7b-2 — 10:40 AM

Perceptual Artifacts in Speech Noise Suppression

Devangi N. Parikh, David V. Anderson, Georgia Institute of Technology

Single-microphone speech enhancement algorithms often exchange a decrease in background noise for an increase in distortion or noise artifacts. We investigate a noise suppression system that is based on a model of human auditory perception and show that suppression artifacts can be significantly reduced by proper control of the temporal evolution of the subband gain signals. This approach is then extended to work with several other speech enhancement algorithms.

MA7b-3 — 11:05 AM

Point Process Models of Spectro-Temporal Modulation Events for Speech Recognition

Aren Jansen, Nima Mesgarani, Johns Hopkins University; Partha Niyogi, University of Chicago

Neurobiological research has uncovered the existence of cortical neurons in various animal species tuned to particular spectro-temporal modulations in the auditory stimulus. Other findings indicate that temporal statistics of the resulting neural spike trains may encode species-specific communication calls. With this motivation, we present an alternative approach to speech recognition based on point process statistical models of the local maxima events produced by a cortically-inspired spectro-temporal filter bank. We demonstrate the computational adequacy of this approach on the practical task of keyword spotting. In conjunction with an unsupervised adaptation strategy, we demonstrate an improved robustness to noise.

MA7b-4 — 11:30 AM

Noise Robust Encoding of Speech in the Primary Auditory Cortex

Nima Mesgarani, Johns Hopkins University

Humans can robustly perceive speech despite considerable variability across speakers and context, and in the presence of acoustic distortions. The neurophysiological basis of these perceptual abilities remains unknown. In this study, we demonstrate how the population of auditory cortical neurons encode the perceptually important features of phonemes along various dimensions. This multidimensional decomposition provides insights into the encoding scheme of complex sounds such as speech by the auditory cortical neurons and suggests alternative ways of analyzing speech. Using a model of this representation enables us to incorporate this knowledge into biologically inspired speech applications that are robust to noise.

Track 1 – A. Communications Systems

Session: MAb8 – Communication Systems I — 10:15 AM – 11:55 AM

Chair: ***Joerg Kliewer,*** *New Mexico State University*

MA8b1-1

Dual Domain Echo Cancellers for Multirate Discrete Multitone Systems

Neda Ehtiati, Benoit Champagne, McGill University

Digital echo cancellers are used in duplex digital subscriber lines to remove the echo. In discrete multitone (DMT) systems, the structure present in the transmitted signal is exploited to reduce the complexity of these cancellers. Previously, we have proposed a novel dual domain echo canceller for symmetric rate DMT systems. This canceller has a faster convergence compared to the existing methods with similar complexity. In this paper, we develop this canceller for multirate DMT systems, and show that by using the polyphase decomposition of the time-domain signals, an efficient implantation for this algorithm can be achieved.

MA8b1-2

Finite Random Matrices for Blind Spectrum Sensing

Giuseppe Abreu, University of Oulu; Wensheng Zhang, Yukitoshi Sanada, Keio University

We address the Primary User (PU) detection (spec- trum sensing) problem, relevant to cognitive applications, from a finite random matrix theoretical (RMT) perspective. Utilizing recently-derived closed-form and exact expressions for the distri- bution of the standard condition number (SCN) of dual random Wishart matrices, we design a new blind algorithm to detect the presence of PU signals. An inherent property of the technique, which is due to the reliance on SCN's, is that no SNR estimation or any other information on the PU signal is required. Like some similar asymptotic RMT-based techniques recently proposed, the algorithm also admits for a tolerated probability of false alarm ▯ to be accounted for by design. The proposed finite RMT-based algorithm, however, outperforms all known similar alternatives, in consequence of the fact that the distribution of SCN's utilized are in closed-form and exact, for any given matrix size.

MA8b1-3

MAP Detection with Soft Information in an Estimate and Forward Relay Network

Corina I. Serediuc, Rice University; Jorma Lilleberg, University of Oulu / Nokia; Behnaam Aazhang, Rice University

One proven solution to improve the reliability of a wireless channel is to use relays. In this paper we will focus our attention on a relay protocol where the relay node estimates source symbol and then forwards it to the destination, called estimate and forward (EF). The EF protocol is characterized by the soft information sent by the relay, which we assume to be the expected value of the symbol transmitted by the receiver. Using bit-error-rate (BER) as a metric, we show that EF outperforms other common relay protocols, in a pathloss system for the cooperative three node wireless network. We also provide a solution to bypass the numerical convolution needed by the MAP detector at the destination.

MA8b1-4

Assisted Radio Field Prediction with Application to Cognitive Radio

Michele Scagliola, Carlos Mosquera, Veronica Santalla del Rio, University of Vigo

Sensing the radio spectrum is an essential feature of cognitive radio. In order to estimate the spatial distribution of the power spectrum, we propose a novel approach combining the informations delivered by a network of sensors and the knowledge of the spatial correlation due to shadowing. If heights and locations of obstacles are available, the spatial correlation of shadowing can be predicted deterministically, so power spectrum, modeled as a random field, can be estimated at a given location by using kriging interpolation.

MA8b1-5

Robust AF Relay Transmission with Multiple Source-Destination Pairs under Channel Uncertainty

Yupeng Liu, Athina Petropulu, Drexel University

The paper considers a scenario where multiple single antenna source-destination node pairs in the network need to communicate simultaneously. An Amplify-and-Forward (AF) relay node equipped multiple antennas assists the communication. The goal of the relay is to satisfy the SINR constraint of each source-destination (S-R) pair and minimize the relay transmit power. Perfect channel state information (CSI) of the relay is required to achieve this goal. However, in practical system, the relay can only obtain a estimate of CSI so that the channel uncertainty will severely reduce the SINR of each S-R pair as well as the performance of outage probability. In this paper we design a low complexity robust relay transmission scheme to combat the channel uncertainty, by 1) computing the maximum SINR loss caused by the channel uncertainty and 2) increasing the SINR threshold of each S-D pair. Simulation results show that the proposed method greatly reduce outage probability. The proposed method employs ZFBF structure and only needs to change each S-D pair's SINR requirement to combat the channel uncertainty thus has low complexity.

MA8b1-6

A Mutual Information based Iteration Stop Rule for Turbo Decoding

Jinhong Wu, Atheros Communications, Inc.; Branimir Vojcic, Jia Sheng, George Washington University

We introduce a new iteration stop rule for turbo codes based on the mutual information between the decoder's hard decision and its soft output.We provide a simple yet effective estimation of the mutual information. The convergence properties of the estimated mutual information are examined. Compared with the well known cross-entropy based stop rule, this method shows higher efficiency.

MA8b1-7

Cooperation Diversity for OFDM with Iterative Reception and Independent CFO per Node

Thomas Ketseoglou, California State University

In this paper, an optimized iterative destination node reception approach is introduced suitable for Orthogonal Frequency Division Multiplexing (OFDM) cooperation diversity. Each of the cooperation-involved nodes presents independent carrier frequency offset (CFO) to the destination receiver. The optimized iterative destination receiver employs the Expectation-Maximization (EM) algorithm to iteratively estimate the doubly selective channels of the source and relay nodes, compensate for the two CFO, and decode the message. Due to the multiplicity of CFO present, the resulting detection algorithm becomes more involved. The system employs M-ary Phase Shift Keying (MPSK) modulation. A minimum number of embedded pilots is used for initial channel estimation of each link through the Lest Squares Estimation (LSE) algorithm. Both systems with and without Automatic Repeat Request (ARQ) are considered. It is shown that high cooperation diversity gains are feasible at the destination, although the system includes multiple sources of uncertainties (including channel parameters, and independent per node CFO).

MA8b1-8

Joint Transmitter Adaptation and Power Control in Multi-User Wireless Systems with Target SIR Requirements

Dimitrie C. Popescu, Shiny Abraham, Old Dominion University; Octavia Dobre, Memorial University of Newfoundland

In this paper we study joint transmitter adaptation and power control in multiuser wireless systems with multipath channels and target values imposed on the users' signal-to-interference+noise-ratio (SINR) at the receiver. We use a general framework that is applicable to a wide range of wireless systems and scenarios to cast this as a distributed constrained optimization problem, and study necessary conditions that must be satisfied by the optimal solution. We also present an incremental algorithm that adapts user transmitters and power values in a distributed fashion until a fixed point is reached where the specified target SINRs are achieved with minimum transmitted power. Convergence of the proposed algorithm is discussed, and the algorithm is illustrated with numerical examples obtained from simulations.

MA8b1-9

Complexity Reduction for Vehicular Channel Estimation Using the Filter-Divergence Measure

Laura Bernadó, Thomas Zemen, FTW Forschungszentrum Telekommunikation Wien; Alexander Paier, Vienna University of Technology; Johan Karedal, Lund University

A key component in vehicular communications systems is the channel estimation filter that suppresses the additive noise in the channel estimates from pilot symbols. A filter which offers the best performance in terms of mean square error (MSE) is the well known Wiener filter. A drawback of using filters based on second order statistics is that they have to be recalculated when the statistical properties of the channel have changed. In vehicular communications the observed channels do not follow the wide-sense stationary (WSS) uncorrelated-scattering (US) properties, and therefore their power spectral density varies over time. A non-stationary process can be divided in time into consecutive stationarity regions where the WSS and US properties are assumed to hold, allowing to calculate the coefficients of a Wiener filter. In this paper we analyze the increase of the MSE observed when using a mismatched Wiener filter. The mismatch results from using the filter coefficients calculated for a past stationarity region. We relate this concept of performance degradation to spectral distance metrics. We use the spectral divergence between

scattering functions at different time instances. Furthermore, we introduce a new metric, the filter divergence, which takes noise into account. We show that, by accepting an increase of MSE, the same filter coefficients can be used for several time regions, which allows computational complexity reduction in a real system.

MA8b1-10
Feasibility and Limitations in Relaying Broadcast Signals
Eun-Hee Shin, Dongwoo Kim, Hanyang University

In this paper, we investigate feasibility and limitations in relaying broadcast signals. We find the angle and radius that can be maximally covered by a single relay while not degrading the original data rate achieved without the relay. During a unit-time slot divided into two phases with different lengths, a source and a relay broadcast data in the first and the second phase, respectively. The sum of transmit power at the source and the relay is constrained. We provide and compare two schemes, the optimal time and power allocation and the optimal power and equal-time allocation with maximum ratio combining at destinations. When the relay can access a fraction of transmit time and power without degrading the original data rate, the relay is feasible. Even with the optimal resource allocation, the coverage of relaying broadcast and the possible location for the relay are limited.

MA8b1-11
An Improved Synchronization Scheme for OFDMA Systems with Initial Ranging Transmissions
Sungeun Lee, Xiaoli Ma, Georgia Institute of Technology

At the initial ranging process, it is crucial to estimate multi-user's multiple parameters such as ranging code, timing and frequency offset and to identify these multiple parameters for each user. This paper presents an improved parameter estimation and pairing algorithm for initial ranging process in OFDMA uplink systems by exploiting multidimensional harmonic retrieval (HR) technique. Unlike most existing techniques that estimate each parameters independently and associate the ranging codes with the estimated parameters manually, the proposed method estimates multiple user's ranging code, timing and frequency offset simultaneously, and pair up all the multiple parameters automatically. The simulation results confirm that the proposed technique not only improves the ranging code detection capability and adjusts the acquisition range of the estimators, but also increases the maximum number of resolvable users for a given samples.

MA8b1-12
Syndrome Based Adaptive Complexity Channel Decoding and Turbo Equalization for ATSC DTV
Klaus Hueske, Jan Geldmacher, Jürgen Götze, TU Dortmund University

Syndrome decoding of trellis based codes is a powerful method to adapt the channel decoding complexity to different transmission conditions in wireless systems. This allows a reduction of the receiver's energy consumption in above-average transmission conditions. Further savings can be achieved if only a specific bit error rate (BER) threshold has to be reached, like the "threshold of visibility" (TOV) or the "quasi error free" (QEF) threshold in the terrestrial ATSC DTV broadcasting system. If Turbo equalization is considered, the syndrome approach enables a reduction of the soft output decoder's complexity, which further decreases during the iteration process.

MA8b1-13
Joint Signal Detection and Classification of Mobile WiMAX and LTE OFDM Signals for Cognitive Radio
Alaa Al-Habashna, Octavia A. Dobre, Ramachandran Venkatesan, Memorial University of Newfoundland; Dimitrie C. Popescu, Old Dominion University

Spectrum awareness is one of the most challenging tasks in cognitive radio (CR). To adequately adapt to the changing radio environment, it is necessary for the CR to be able to perform joint detection and classification of low signal-to-noise ratio (SNR) signals without requiring much a priori information on the signal parameters. In this paper, we propose a joint detection and classification algorithm for the mobile Worldwide Interoperability for Microwave Access (WiMAX) and Long Term Evolution (LTE) signals. The algorithm has the advantage that it requires relaxed information on the signal parameters. Simulation results are presented, which show the efficiency of the proposed algorithm under diverse scenarios.

MA8b1-14
Cooperative Game-Theoretic Solutions to Spectrum Sharing in Cognitive Radios
Jayaprakash Rajasekharan, Jan Eriksson, Visa Koivunen, Aalto University

This paper addresses the problem of spectrum sharing in cognitive radios where secondary users cooperatively sense the spectrum for identifying and accessing unoccupied spectrum bands. It is shown that this scenario can be modeled as a cooperative game to allocate spectrum resources fairly to each user. Users form coalitions to jointly sense and share the spectrum. The worth of each user is calculated according to the work done for the coalition in terms of the information obtained about primary user activity from sensing the spectrum. The resulting games are balanced, thereby ensuring non-empty cores for allocating resources to users. The games are also convex, thus enabling one-point solution concepts such as the Shapley value, tau value and nucleolus to lie within the core to provide stability. The concept and results are illustrated using a simple toy example.

MA8b1-15
Calibration of Random Phase Rotation for Multi-Band OFDM UWB Signals
Huilin Xu, Liuqing Yang, University of Florida

In order to achieve high accuracy ToA estimation, channel information from subbands of the multi-band system needs to be coherently combined. This requires the calibration of the random phase rotation induced by the carrier demodulation of subband signals. Analysis in this paper reveals that presence of the phase rotation reduces the resolution of multi-band signals, which results in more severe energy leakage in the time domain channel estimate. Based on this, the phase rotation can be readily calibrated by mitigating the energy leakage. Simulations are also provided to corroborate the analysis.

MA8b1-16
Wideband Spectrum Sensing for Cognitive Radios in Unknown Noise via Power Spectrum Analysis
Jitendra Tugnait, Auburn University

One of the first steps to be accomplished by a cognitive user in cognitive radio applications is spectrum sensing: analysis of the received electromagnetic transmissions to search for unoccupied spectrum bands (spectrum holes). We propose several spectrum sensing methods based on comparing the power spectral densities estimated from data acquired over two non-overlapping time segments. The first window serves as a reference for noise and interference while the second window may or may not contain primary user(s). Several binary hypothesis tests based on the generalized likelihood ratio test (GLRT) paradigm are formulated for comparing the power spectra estimated from the two time segments. We also investigate approaches based on just one time segment. The approaches are illustrated via simulations.

MA8b1-17
Fair Resource Allocation for Hybrid FSO/RF Networks
Yi Tang, Maite Brandt-Pearce, University of Virginia

Adding optical transceivers to fixed RF nodes promises to significantly enhance the total network throughput. Yet, the free space optical (FSO) subsystem is limited as it comprises only a small number of point-to-point link, each susceptible to fading due to weather and atmospheric turbulence. The RF subsystem suffers from Rayleigh fading and interference. In this paper we formulate and solve for the optimal allocation of lasers link and RF power so that the transmission rate over the network is maximized subject to fairness constraints. The joint system is shown to significantly surpass the individual subsystems in its ability to accommodate fairness.

MA8b1-18
MCM OFDM Using Sparse Signals
Victor DeBrunner, Florida State University; Jim Schroeder, Harris Corporation

We describe our initial approach to MCM OFDM that uses entropy based signal models to provide a sparse, efficient set of codes. Specifically, we consider applying a set of sparse orthogonal signals developed precisely to solve the space-time uncertainty of the Hirschman Uncertainty (itself a generalization of the Heisenberg Uncertainty) for finite-duration discrete time signals. These signals were in part developed by one of the authors, and primarily used in both linear and non-linear adaptive filtering. Recently, that author has begun to use those elements to develop high-resolution spectral estimation methods for the stationary environment. In this paper, we begin to apply these results to one of the most basic communications strategies.

Track 8 – H. Speech, Image and Video Processing

Session: MAb8 – Selected Topics in Image Processing 10:15 AM – 11:55 AM

Chair: ***Christopher Rozell,*** *Georgia Institute of Technology*

MA8b2-1

Sparse Coding for Spectral Signatures in Hyperspectral Images

Adam Charles, Georgia Institute of Technology; Bruno Olshausen, University of California, Berkeley; Christopher Rozell, Georgia Institute of Technology

The growing use of hyperspectral imagery lead us to seek automated algorithms for extracting useful information about the scene. Recent work in sparse approximation has shown that unsupervised learning techniques can use example data to determine an efficient dictionary with few a priori assumptions. We apply this model to sample hyperspectral data and show that these techniques learn a dictionary that: 1) contains a meaningful spectral decomposition for hyperspectral imagery, 2) admit representations that are useful in determining properties and classifying materials in the scene, and 3) forms local approximations to the nonlinear manifold structure present in the actual data.

MA8b2-2

Distributed Compressed Sensing of Hyperspectral Images via Blind Source Separation

Mohammad Golbabaee, Simon Arberet, Pierre Vandergheynst, Ecole Polytechnique Fédérale de Lausanne

This paper describes a novel framework for compressive sampling (CS) of multichannel signals that are highly dependent across the channels. In this work, we assume few number of sources are generating the multichannel observations based on a linear mixture model. Moreover, sources are assumed to have sparse/compressible representations in some orthonormal basis. The main contribution of this paper lies in 1) rephrasing the compressed sampling of multichannel data as a compressive blind source separation problem, and 2) proposing an optimization problem and a recovery algorithm to estimate both the sources and the mixing matrix (and thus the whole data) from the compressed measures. A number of experiments on the acquisition of Hyperspectral images show that our proposed algorithm obtains a reconstruction error by between 10 dB and 15 dB less than other state-of-the-art CS methods.

MA8b2-3

Automatic Feature Extraction in Laser Rangefinder Data Using Geometric Invariance

Jean-Charles Noyer, Régis Lherbier, Benoit Fortin, Univ. Lille Nord-de-France

This paper presents a feature extraction method in scanning laser rangefinder data. Whereas many popular methods use Cartesian coordinates to detect features, the proposed method use a geometric invariant in natural (polar) coordinates. It leads to a membership condition that only depends on the sensor properties (angular resolution and range measurement error).

MA8b2-4

A Novel Facial Expression Recognition Method Using Fast BEMD Based Edge Detection

James Zhang, Zijing Qin, Peter Tay, Robert Adams, Western Carolina University

Traditionally, human facial expressions are recognized using standard images. These methods of recognition require subjective expertise and high computational costs. This article presents a novel facial expression recognition method using edge maps generated from standard images that can significantly improve computational efficiency. The edge maps are generated from a novel Bi-dimensional Empirical Mode Decomposition (BEMD) based edge detection method. In this paper, the BEMD edge detection algorithm is discussed, the facial expression decision metrics are developed, and detection results of facial expression databases are presented. The success rates of recognition suggest that the proposed method be a potentially efficient method for human facial expression detection and other similar recognition applications.

MA8b2-5

Plenoptic Rendering on GPUs

Todor Georgiev, Andrew Lumsdaine, Georgi Chunev, Adobe

Traditional photography constructs the image optically, at the time of capture. Instead, we se split the light into tens of thousands of little 4D volumes recording individual aspects, like parallax angle, defocus, HDR. Then we computationally put together an image, after the fact. The large amount of data is handled on the GPU in real time. Effectively, optics is transferred into the digital domain. This summarizes the result of several years of research. A working demo will be shown.

MA8b2-6
Complexity and Quality Evaluation of Structure Extrapolation Methods Within a Fully Automatic Inpainting Framework
Patrick Ndjiki-Nya, Dimitar Doshkov, Martin Koeppel, Thomas Wiegand, Fraunhofer Institute for Telecommunications - Heinrich-Hertz-Institut

In this work, a previous presented inpainting method is used for the evaluation of different, new structure extrapolation strategies. The important issue of quality-complexity optimization is discussed and efficient inpainting tools are established. Recovery of missing dominant structures is done via tensor voting, regression analysis or a combination of both. Combined structure extrapolation considerably reduces the complexity of structure restoration, compared to genuine tensor voting, with overall graceful to imperceptible degradation of the inpainting result. Furthermore, limitations of the tensor voting approach can be avoided by means of polynomial interpolation with additional savings in terms of complexity.

MA8b2-7
Multi-modal Image Fusion using Window-based ICA and Fractal Dimension
Lu Han, North Carolina State University; Shubha Kadambe, Rockwell Collins Company; Hamid Krim, North Carolina State University

The goal of multi-modal image fusion is to combine complementary information from multisensory data such that the fused image is more suitable for the purpose of human visual perception, computer-processing tasks and detection applications. In this paper, we first use independent component analysis (ICA) as the primary transformation to obtain adaptive analysis bases trained by similar reference images; we then utilize Fractal Dimension (FD) to detect textural information and an intensity histogram to sense objects of interest as a new additional fusion rule. Substantiating examples show an improved performance in perception and in an enhancement of textural information and objects of interest.

MA8b2-8
A Regularized Optimization Approach for AM-FM Reconstructions
Paul Rodriguez, Pontificia Universidad Catolica del Peru; Victor Murray, Marios S. Pattichis, University of New Mexico

The AM-FM Dominant and Channelized Component Analysis (DCA and CCA respectively) [1], consist of applying a filter bank to the Hilbert-tranformed image, and then proceeding with the AM-FM demodulation of each band-pass filtered image. Whereas AM-FM reconstructions based on the CCA use a reasonably small number of locally coherent components, those based on the DCA only use one component: the estimates from the channel with the maximum amplitude estimate. Both types of reconstructions are known to produce noticeable visual artifacts. We propose a method, based on a regularized optimization of the estimates from the CCA, which attains a small number of locally coherent components and simultaneously enforces a piecewise smooth constrain for the amplitude functions. Moreover, this method offers high quality reconstructions when compared to standard CCA and DCA reconstructions and state of the art techniques [2].

MA8b2-9
Block Based Completion for Video Stabilization
Stephen Mangiat, University of California, Santa Barbara; Yi-Jen Chiu, Intel Corporation

Video stabilization helps eliminate unwanted camera motion, yet it also introduces missing regions near the edge of the frame. We outline a new method for the completion of stabilized video using a block based approach. Building upon macroblock recovery techniques, candidate blocks are chosen using the nearby motion vectors of blocks at the edge of the original frame. Then, the candidate block with the best boundary match is used to fill in the missing region. Results on videos of dynamic scenes show effective completions, with the potential to leverage existing block motion estimation hardware and significantly decrease processing time when compared to other state of the art completion methods.

MA8b2-10
ρ-Domain Rate Control for JPEG XR
Duncan Chan, Jie Liang, Simon Fraser University; Chengjie Tu, Microsoft Corp.

A rate control algorithm is introduced to the JPEG XR still image compression algorithm, thereby allowing the user to compress an image to a target bit-rate. Using ρ-domain analysis, we showed a strong linearity relationship of the rate function $R(\rho)$. This allows us to accurately estimate the rate-distortion function, and thus control bit-rate of the JPEG XR encoder.

Session: MAb8 – Applications of Compressive Sensing 10:15 AM – 11:55 AM

Chair: ***Sergiy Vorobyov,*** *University of Alberta Edmonton, Canada*

MA8b3-1

Empirical Risk Minimization-Based Analysis of Segmented Compressed Sampling

Omid Taheri, Sergiy Vorobyov, University of Alberta

A new segmented compressed sampling (CS) method for analog-to-information conversion (AIC) has been proposed in our recent work. Its essence is to collect a larger number of samples (although correlated) than the number of parallel branches of mixers and integrators in the AIC devise. The objective of this paper is to prove that the additional samples obtained based on the proposed segmented CS method lead to improved signal recovery quality. The study is performed based on the empirical risk minimization recovery method, but the least absolute shrinkage and selection operator algorithm can also be viewed as a particular realization of the empirical risk minimization method.

MA8b3-2

Localization in Wireless Networks via Spatial Sparsity

Sofia Nikitaki, Panagiotis Tsakalides, University of Crete & FORTH-ICS

This paper exploits recent developments in sparse approximation and compressive sensing to efficiently perform localization in wireless networks. Particularly, we re-formulate the localization problem as a sparse approximation problem using the compressive sensing theory that provides a new paradigm for recovering a sparse signal solving an l_1 minimization problem. The proposed received signal strength-based method does not require any time specific/propriatery hardware since the location estimation is performed at the Access Points (APs). The experimental results show that our proposed method, when compared with traditional localization schemes results in a better accuracy in terms of the mean localization error.

MA8b3-3

Joint Typical Analysis for Compressive Sensing Based Multi Sensor Systems

Sangjun Park, Junho Lee, Heungno Lee, Gwangju Institute of Science and Technology (GIST)

We investigate the possibility of using compressive sensing (CS) [1] in Multiple Sensor Systems (MSS). In the proposed MSS, we envision that, sensors take measurements while compressing the signal with linear projection operation by means of CS. In this paper, our focus is twofold, (1) to obtain a bound for per-sensor measurements (PSM) required for full recovery of signal at each sensor and (2) to observe how the probability of successful recovery is affected by SNR and the number of sensors. We extend the concept of joint typicality proposed by Akçakaya and Tarokh[2] to this MSS problem. Our result is that PSM converges to the sparsity—the number of non-zero elements—as the number of sensors increases. This interesting result, we note, is consistent with the previous results by Sarvotham et al.[3] and JSM-2 Model of Baron et al.[4]; but our result is more general because the effect of noise is considered in our work.

MA8b3-4

Compressive Imaging using Approximate Message Passing and a Markov-Tree Prior

Subhojit Som, Lee C Potter, Philip Schniter, Ohio State University

We propose a novel compressive imaging framework that exploits the persistence-across-scales property exhibited by the (sparse) wavelet coefficients of natural images. In particular, we model wavelet coefficient sparsity using a hidden Markov tree, and we estimate the coefficient values from measurements taken at sub-Nyquist rates using the framework of belief propagation, leveraging recent work in approximate message passing (AMP). The proposed algorithm can be interpreted as a form of turbo sparse reconstruction, where extrinsic beliefs on the coefficient support are exchanged by two soft inference blocks, one exploiting structure in the sensing channel and the other exploiting structure in the Markov prior. We compare this algorithm to other recently proposed compressive imaging methods that leverage the persistence-across-scales property.

MA8b3-5

Computable Quantification of the Stability of Sparse Signal Reconstruction

Gongguo Tang, Arye Nehorai, Washington University in St. Louis

The ℓ_1-constrained minimal singular value (ℓ_1-CMSV) of the sensing matrix is shown to determine, in a concise and tight manner, the recovery performance of ℓ_1-based algorithms such as Basis Pursuit, the Dantzig selector, and the LASSO estimator. Several random measurement ensembles are shown to have ℓ_1-CMSVs bounded away from zero with high probability, as long as the number of measurements is relatively large. Three algorithms based on projected gradient method and interior point algorithm are developed to compute ℓ_1-CMSV. A lower bound of the ℓ_1-CMSV is also available by solving a semi-definite programming problem.

MA8b3-6
Signal Recovery from Low Frequency Components
Yonina C. Eldar, Technion - Israel Institute of Technology; Volker Pohl, Technical University Berlin

In many applications only the low frequency components of a signal can be measured due to the lowpass behavior of many physical systems. Nevertheless, if additional information on the structure of the signal is known, it might still be possible to reconstruct the signal from its low-frequency content. This paper studies signals in shift-invariant spaces with multiple generators and derives necessary conditions on the bandwith of the lowpass filter as well as sufficient conditions on the generators such that signal recovery is possible. If the signal can not be recovered from its low frequency components, an appropriate pre-processing of the signal is proposed which improves the reconstruction ability. In particular, we will show that modulating the signal with one or more mixing functions prior to lowpass filtering can ensure the recovery of the signal in many cases.

Track 1 – A. Communications Systems

Session: MPa1 – Interference Channels

Chair: ***Eduard Jorswieck,*** *Technische Universität Dresden*

MP1a-1 1:30 PM
Learning to Precode in Outage Minimization Games over MIMO Interference Channels
Elena Veronica Belmega, Hamidou Tembine, Samson Lasaulce, Laboratoire des signaux et systèmes

The type of networks considered in this paper consists of a 2-transmitter 2-receiver interference channels for which all the terminals are equipped with multiple antennas. Based on a reduced feedback, the objective of each transmitter is to choose its precoding scheme in order to minimize its individual outage probability. This problem is modeled as a game which is assumed to be played a large number of times. It is shown how learning algorithms can be used to obtain equilibria in this game. The efficiency of the latter is analyzed and an algorithm is proposed to improve the performance of the global system.

MP1a-2 1:55 PM
Achievable Rates and Upper Bounds for the Interference Relay Channel
Anas Chaaban, Aydin Sezgin, Ulm University

The two user Gaussian interference channel with a full-duplex relay is studied. By using genie aided approaches, two new upper bounds on the achievable sum-rate in this setup are derived. These upper bounds are shown to be tighter than previously known bounds under some conditions. Moreover, a transmit strategy for this setup is proposed. This strategy utilizes the following elements: Block Markov encoding combined with a Han-Kobayashi scheme at the sources, decode and forward at the relay, and Willems' backward decoding at the receivers. This scheme is shown to achieve within a finite gap our upper bounds in certain cases.

MP1a-3 2:20 PM
Bargaining and Beamforming in Interference Channels
Rami Mochaourab, Eduard A. Jorswieck, Dresden University of Technology; Zuleita Ka Ming Ho, David Gesbert, Eurecom

Utilizing the real-valued parameterization of each transmitter's efficient beamforming vectors, we propose a decentralized resource allocation scheme in the multiple-input single-output interference channel. The scheme is motivated by bargaining concepts in game theory. The aim of these concepts is to improve the joint payoff of the users from the Nash equilibrium outcome. In each bargaining stage, each user proposes a strategy. A user accepts any proposal if it increases his payoff. Otherwise, new proposals are made. When all proposals are accepted, a new stage begins. We prove the scheme's convergence and demonstrate its performance by simulations. In comparison to previous approaches, our bargaining outcome is arbitrarily close to the Pareto boundary of the achievable single-user rate region. We further discuss the control overhead and complexity of this scheme.

MP1a-4 2:45 PM

Optimal Distributed Beamforming for MISO Interference Channels

Jiaming Qiu, Texas A&M University; Rui Zhang, National University of Singapore; Zhi-Quan Luo, University of Minnesota; Shuguang Cui, Texas A&M University

We consider the problem of quantifying the Pareto optimal points in the achievable rate region over MISO interference channels, where the problem boils down to a sequence of convex feasibility problems after certain transformations. The feasibility problem is solved by two distributed beamforming algorithms , where the first one is based on the convex alternating projection method, and the second one is based on cyclic projection method. Convergence proofs are established for both algorithms.

Track 2 – B. MIMO Communications and Signal Processing

Session: MPa2 – MIMO Secrecy

Chair: ***A. Lee Swindlehurst,*** *University of California, Irvine*

MP2a-1 1:30 PM

Secrecy in Gaussian MIMO Bidirectional Broadcast Wiretap Channels: Transmit Strategies

Sara Al-Sayed, Aydin Sezgin, Ulm University

We examine transmit strategies for secrecy in the Gaussian MIMO Bidirectional Broadcast Wiretap Channel. We first consider the case with perfect knowledge of the eavesdropper channel at the transmitter. Afterwards, we investigate the impact of imperfect knowledge of the eavesdropper channel including the extreme case of having no information on the eavesdropper. With the eavesdropper present, the transmitter has to adopt the transmit strategy in order to fulfill given secrecy constraints. This adoption, however, results in a loss in terms of achievable rates. An upper bound on this loss is derived and shown to be reasonably tight for a huge range of SNR values. In the absence of perfect channel knowledge of the eavesdropper's channel, artificial noise broadcasting is considered. Finally, the theoretical results are illustrated by means of numerical simulations.

MP2a-2 1:55 PM

Maximization of Worst-Case Secrecy Rates in MIMO Wiretap Channels

Anne Wolf, Eduard A. Jorswieck, Dresden University of Technology

We study a multi-antenna wiretap channel, where the transmitter does not have perfect knowledge about the channels to the eavesdropper. The transmitter only knows that the logical location of the eavesdropper is drawn from a certain set. We characterize the worst-case secrecy rate in this scenario for given transmit strategies and derive an upper and a lower bound for the worst-case secrecy rate that is maximized under a sum power constraint at the transmitter. Finally, we discuss strategies for the high and low SNR regime and illustrate our results.

MP2a-3 2:20 PM

Ergodic Secrecy Rate for Gaussian MISO Wiretap Channels with Rician Fading

Jiangyuan Li, Shuangyu Luo, Athina Petropulu, Drexel University

A Gaussian multiple-input single-output (MISO) wiretap channel model is considered, where there exists a transmitter equipped with multiple antennas, a legitimate receiver and an eavesdropper each equipped with a single antenna. We study the problem of finding the optimal input covariance that achieves ergodic secrecy rate subject to a power constraint where the full information on the legitimate channel is known to the transmitter, but only statistical information on the eavesdropper channel is available at the transmitter. Existing results address the case in which the eavesdropper channel has independent and identically distributed Gaussian entries with zero-mean, i.e., the channel has trivial covariance. This paper addresses the general case where eavesdropper channel has non-zero mean and nontrivial covariance. We show that the optimal input covariance has always rank one. Based on this, for the non-zero mean but trivial covariance, we reduce the original problem to an one variable optimization problem and propose Newton's method to solve the resulting optimization problem. Numerical results are presented to illustrate the algorithm.

MP2a-4 2:45 PM

Robust Beamforming for MISO Wiretap Channel by Optimizing the Worst-case Secrecy Capacity

Wei Shi, James Ritcey, University of Washington

A Gaussian multiple-input single-output (MISO) secrecy channel is considered, where a transmitter is equipped with multiple antennas, and an intended receiver and an eavesdropper each equipped with a single antenna. This paper studies the robust beamforming design in the presence of uncertainty of channel state information (CSI) on the eavesdropper channel. Under the assumption that all the possible states of the eavesdropper channel are given, the problem of optimizing the input covariance for the worst-case secrecy capacity which is the maximum transmission rate at which the eavesdropper is unable to decode under any channel realization, is cast as a max-min optimization problem. This max-min problem is non-convex and is generally difficult to solve. We propose to transform the max-min problem into a convex optimization problem that can be solved efficiently. We show that the optimal input covariance is rank-one, i.e. beamforming optimal. The optimal beamforming vector can be expressed in the form of the principle generalized eigenvector, the same form as the conventional case of the perfect eavesdropper's CSI, which indicates the conventional case is a special case of our work.

Track 3 – C. Networks

Session: MPa3 – New Trends in Information Theory and Networks

Co-Chairs: ***Sanjay Shakkottai,*** *University of Texas at Austin* ***and Jeff Andrews,*** *University of Texas at Austin*

MP3a-1 1:30 PM

On Information Theoretic Games for Interference Networks

Suvarup Saha, Randall Berry, Northwestern University

Game theory provide a useful framework for studying the interactions of multiple autonomous users in an interference network. This paper builds on recent work that establishes an information theoretic model for such games in which users can choose any encoding and decoding strategy to maximize their own rate. While previous work has focused mainly on two user interference channels, here we consider models with more than two users.

MP3a-2 1:55 PM

Correlation of Link Outages in Low-mobility Wireless Networks

Radha Krishna Ganti, Jeffrey Andrews, University of Texas at Austin

The notion of link outages in a wireless network depends on the time-scales involved, and in the literature all works assume either high-mobility of the transmitters or obtain the spatial average of the outage, assuming ergodicity. But for a fixed locations of nodes, each link encounters a different outage probability which can significantly differ from the average outage probability. In this paper we consider a network where the location of the nodes is drawn from a given spatial distribution and remains fixed (at least for a long duration of time). The link outage probability depends on the transmit-receive pair location, fading and we derive the distribution of these link outage probabilities. This work supplements the existing results which assume that both the spatial locations and fading change (resampling) for every packet transmitted.

MP3a-3 2:20 PM

On Information Utility and Generalization of Data Processing Inequality

Tara Javidi, University of California, San Diego

This paper connects the stochastic control theoretic notion of information utility to the concept of uncertainty reduction in information theory. In particular, the paper first provides a brief survey of the design of experiment literature and the dynamic programming interpretation of information utility introduced by De Groot. Then it is shown that these notions and results form a basis for natural extensions of mutual information as well as data processing inequality. Finally, these extensions are contrasted with Csiszar's measures of informativity of observation channels defined by the aid of f-divergence (as a generalized measure of difference of two probability distributions).

MP3a-4 2:45 PM

On the Significance of Linear Codes in Networks

Jiening Zhan, Michael Gastpar, University of California, Berkeley

We discuss the properties and importance of linear codes in communication networks. We explore various aspects, including capacity and synchronization problems.

Track 6 – F. Biomedical Signal and Image Processing

Session: MPa4 – Biomedical Image Analysis

Chair: ***Ronald Summers,*** *National Institutes of Health*

MP4a-1 1:30 PM

Haustral Fold Detection for CT Colonography Images Using Gabor Filter

Zhuoshi Wei, Jianhua Yao, Shijun Wang, Ronald Summers, National Institutes of Health

Haustral folds are anatomically meaningful structures in the human colon. In this paper we present a method for haustral fold detection for CT colonography images. The 3D colon surface is first preprocessed into a 2D flattened colon. Then the image processing is done on 2D images. The proposed method makes used of a 2D Gabor filter to extract the feature of haustral folds. Then Sobel operator is performed on the filtering results to acquire the edge of the folds. Finally the folds are identified using thresholding method. Experimental results show that our method has a satisfactory detection rate on haustral folds.

MP4a-2 1:55 PM

Human Activity Recognition Via Motion and Vision Data Fusion

Chun Zhu, Qi Cheng, Weihua Sheng, Oklahoma State University

Automated recognition of human daily activities is very important for human-robot interaction (HRI) in assisted living systems. We propose a Bayesian framework to integrate motion sensor observations and the location information from a vision system for human daily activity recognition. Two problems are studied in this paper: enhancing activity recognition through the fusion of two channels of information and learning the environment through the activity distribution map. The entropy associated with human activity recognition is adopted as an evaluation metric in both problems. The simulation results demonstrate the feasibility of the proposed methods.

MP4a-3 2:20 PM

Segmentation and Pseudo-coloring of High-Speed Bright-Field Microscopy Images of the Beating Embryonic Heart

Sandeep Bhat, Michael Liebling, University of California, Santa Barbara

High-speed, bright-field microscopy image sequences of the beating embryonic zebrafish heart reveal the motion of cardiac tissue, red blood cells, as well as static background structures. Using a dynamic programming and wavelet-based separation algorithm that relies on the periodic, aperiodic, and static motion patterns respective to each of these structures, we report on our progress to extract their contributions into three separate channels, thereby improving upon a previously developed algorithm that was limited to two-class segmentation only. We show that the three extracted channels can be recombined into a pseudo-colored sequence to offer fluorescence microscopy-like visualization and the ability to analyze the dynamic characteristics of individual cardiac structures at frame-rates not achievable with fluorescence microscopy. We expect these improvements to facilitate quantitative characterization of heart function during normal and abnormal cardiac development.

MP4a-4 2:45 PM

A Scheme of Bandwidth Allocation for the Transmission of Medical Data

Di Lin, Fabrice Labeau, McGill University

In this paper, a bandwidth allocation scheme for the transmission of medical data in the IEEE 802.11n based WLAN is proposed. The main idea is to transmit medical data on the subcarriers for realtime traffic when these subcarriers are available and, otherwise, to store these medical data in patient devices; this is a tradeoff between the traffic load in the WLAN and the memory size of patient devices. In addition, based on the patient status, the medical data are classified into normal data and abnormal data, and the delay requirement of the transmission of abnormal medical data is taken into account. Based on this scheme as well as the requirements of various types of traffic, the method for optimal bandwidth allocation to minimize the number of subcarriers is obtained by solving a dynamic programming problem.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: MP5 – Statistical Signal Processing for Complex Systems

Chair: ***Monica Bugallo,*** *SUNY Stony Brook*

MP5-1 1:30 PM

Likelihood Consensus: Principles and Application to Distributed Particle Filtering

Ondrej Hlinka, Ondrej Sluciak, Franz Hlawatsch, Institute of Communications and Radio-Frequency Engineering, Vienna University of Technology; Petar Djuric, Stony Brook University; Markus Rupp, Institute of Communications and Radio-Frequency Engineering, Vienna University of Technology

We propose a distributed method for computing the joint (all-sensors) likelihood function in a wireless sensor network. A consensus algorithm is used to calculate a sufficient statistic describing the (approximate) joint likelihood function. This ``likelihood consensus'' method requires only communications between neighboring sensors. We implement the likelihood consensus method in a distributed particle filtering scheme using local low-rate communications. Each sensor runs a local particle filter that estimates the global state. The updating of the particle weights at each sensor exploits the joint likelihood function, which epitomizes the measurements of all the sensors. This approach is demonstrated for a target tracking problem.

MP5-2 1:55 PM

Particle Filtered Modified Compressive Sensing (PF-mod-CS) for Tracking Signal Sequences

Samarjit Das, Namrata Vaswani, Iowa State University

We introduce a new approach for Bayesian tracking of signal sequences over time. A key limitation of the recently proposed PF-MT is that it needs to know the set of ``dominant" signal change directions (in a given basis). In many situations this set may not be known, and it often changes with time, though slowly. In other words, the signal changes are often approximately sparse in a certain basis (referred to as the ``sparsity basis" in compressive sensing (CS) literature), and the sparsity pattern changes slowly over time. This interpretation allows us to use modified-CS for this purpose.

MP5-3 2:20 PM

Compressed Sensing using Generalized Polygon Samplers

Kanke Gao, Stella Batalama, Dimitris Pados, State University of New York at Buffalo; Bruce Suter, Air Force Research Laboratory

We propose new deterministic low-storage constructions of compressive sampling matrices based on finite-geometry generalized polygons. For the noiseless measurements case, we develop a novel exact-recovery algorithm for strictly sparse signals that utilizes the geometry properties of generalized polygons and exhibits complexity linear in the sparsity value. In the presence of measurement noise, recovery of the generalized-polygon sampled signals can be carried out effectively using a belief propagation algorithm.

MP5-4 2:45 PM

Particle Filtering with Mode Tracking: Potential for Application to Numerical Weather Prediction

Sarah Dance, Joanne Pocock, Amos Lawless, University of Reading

An essential part of weather forecasting is to use observations to estimate the state of the atmosphere. Weather prediction models are highly nonlinear; this might encourage use of the particle filter for state estimation. However, there are problems with implementation of the particle filter for high dimensional systems. Here we compare the standard particle filter (PF) with Vaswani's mode-tracking particle filter (PF-MT) that was designed to reduce the required sample size. We find that PF-MT outperforms PF when a perfect model is used as forecast model. However, when a biased model is used, the performance of PF-MT is degraded.

BREAK 3:10 PM

MP5-5 3:30 PM

Compressible Priors for Natural Image Statistics

Volkan Cevher, EPFL STI IEL LIONS

We describe a set of probability distributions, dubbed compressible priors, whose independent and identically distributed (iid) realizations result in p-compressible signals. A signal x called p-compressible if its magnitude sorted coefficients exhibit a power-law decay, where the decay rate is equal to 1/p. We show via experiments that the wavelet coefficients of natural images are 1.67-compressible whereas their pixel gradients are 0.95log(N/0.95)-compressible, on the average. Based on this observation, we discuss how to exploit compressible priors for denoising and deblurring of natural images.

MP5-6 3:55 PM

Unsupervised Bayesian Analysis for Gene Expression Analysis

Cécile Bazot, Nicolas Dobigeon, Jean-Yves Tourneret, University of Toulouse; Alfred O. Hero, University of Michigan

In the past few years, genomics has received growing interest in the signal processing community. However, medical teams are currently facing a new challenge: processing the signals issued by DNA chips. These signals allow one to discover the level of a gene expression in a given tissue at any time. This paper introduces a new method to identify molecular signatures involved in symptomatic disease from the data issued by these DNA chips. The problem is formulated as a constrained matrix factorization problem that is solved using a Bayesian algorithm. The complexity of the Bayesian estimators is alleviated by a Monte Carlo-based procedure providing point estimates as well as confidence intervals for these estimates.

MP5-7 4:20 PM

Multiple Sensor Sequential Tracking of Neural Activity: Algorithm and FPGA Implementation

Lifeng Miao, Jun Zhang, Chaitali Chakrabarti, Antonia Papandreou-Suppappola, Arizona State University

In this paper, we propose to track neural activity by efficiently estimating the locations and moments of multiple dipole sources using noninvasive magnetoencephalography (MEG) measurements from multiple sensors. Our approach is based on reducing implementation complexity by using the multiple particle filter estimation tracker and by implementing each subspace particle filter in parallel using the independent Metropolis-Hastings algorithm to reduce processing time. We also investigate a parallel implementation mapping of the algorithm on a Xilinx Virtex-5 field-programmable gate array (FPGA) platform. Simulation results demonstrate improvements in mean-squared tracking error performance as well as significant reductions in timing requirements of the parallel hardware implementation for real time applications.

MP5-8 4:45 PM

Adaptive Parameter Estimation of Cardiovascular Signals Using Sequential Bayesian Techniques

Shwetha Edla, Jun Jason Zhang, John Spanias, Antonia Papandreou-Suppappola, Chaitali Chakrabarti, Arizona State University

Parameter estimation of electrocardiogram (ECG) signals can be used to monitor cardiac health and diagnose heart diseases. However, statistical ECG models with unknown parameters depend upon a priori and user-specified parameters such as mean cardiac frequency that can vary from patient to patient and at different disease stages. We propose a sequential Bayesian tracking method to adaptively select cardiac parameters at each time step in order to minimize estimation errors. Our results using real ECG data demonstrate the importance of the adaptive algorithm for selecting cardiac parameters and show how these parameters can be used to classify different types of ECG signals.

Track 7 – G. Architecture and Implementation

Session: MP6 – Communication Processors and Accelerators

Chair: ***J. Cavallaro,*** *Rice University*

MP6-1 1:30 PM

Reconfigurable MIMO Transceiver Design using the Tunable Channel Decomposition

Jing Wang, Gerald Sobelman, University of Minnesota

Available spectrum resources are becoming scarcer, while additional services are being integrated into wireless devices. Cognitive radio and multiple-antenna (MIMO) techniques can be combined to provide a solution to this dilemma. Given the currently available frequency bands and a set of bit-rate constrained services to be transmitted, the antennas can be partitioned among frequency bands and the recently proposed tunable channel decomposition (TCD) may be applied to obtain subchannels having prescribed gains. We present a reconfigurable closed-loop transceiver design which can adapt to various partitioning scenarios. As a result, our system can deliver the required services with reduced transmitting power.

MP6-2 1:55 PM

A WiMAX/LTE Compliant FPGA Implementation of a High-Throughput Low-Complexity 4x4 64-QAM Soft MIMO Receiver

Vadim Smolyakov, Dimpesh Patel, University of Toronto; Mahdi Shabany, University of Toronto / Sharif University; Glenn Gulak, University of Toronto

This paper presents a prototype of a highthroughput 4x4 64-QAM MIMO receiver consisting of a channel matrix QR decomposition, a soft-output KBest MIMO detector and a Convolutional Turbo Code decoder. The proposed MIMO receiver provides low processing latency and a pipelined architecture scalable to a larger number of antennas and constellation order. Therefore, it is suitable for LTE-Advanced and IEEE 802.16m broadband wireless standards. A rapid prototyping platform interfacing MATLAB with Xilinx ISE was used in the development of the 4x4 64-QAM MIMO receiver. The receiver utilizes 96% of the slice LUTs and 78% of slice registers on Virtex-5 FX130T FPGA, operating at a maximum frequency of 125MHz.

MP6-3 2:20 PM

An Ultra Low Power SIMD Processor for Wireless Communications

Mark Woh, Sangwon Seo, Scott Mahlke, Trevor Mudge, University of Michigan; Chaitali Chakrabarti, Arizona State University

This paper presents an ultra low power programmable processor architecture for baseband processing in wireless communications. It is a wide-SIMD architecture where the SIMD width can be configured at run time to the specifics of the algorithm being executed. A study of the memory organization and data layout to support the different memory access patterns will be presented. Analysis of the effect of process variation on ultra low voltage operation of the proposed wide-SIMD architecture will be shown. Algorithm and circuit techniques to contain the variation in the SIMD data path will also be presented.

MP6-4 2:45 PM

Combined Channel and Hardware Noise Resilient Viterbi Decoder

Amr Hussien, Muhammed Khairy, Amin Khajeh, Ahmed Eltawil, Fadi Kurdahi, University of California, Irvine

Recent power reduction techniques aggressively modulate the supply voltage of embedded buffering memories allowing acceptable hardware errors to flow through the processing chain. In this paper, we present a model that captures the statistics of both channel noise and hardware failures. We further introduce a modified Viterbi decoder that maximizes the likelihood of the received data based on the distribution of the combined noise. Simulation results show a consistent improvement in BER performance across all SNRs with a area overhead ranging from 0.65% to 3.26% compared to the conventional Viterbi decoder when synthesized using a 65 nm standard library.

BREAK 3:10 PM

MP6-5 3:30 PM

Implementation of Greedy Algorithms for LTE Sparse Channel Estimation

Patrick Maechler, Pierre Greisen, Benjamin Sporrer, Sebastian Steiner, Norbert Felber, Andreas Burg, ETH Zurich

Broadband wireless systems often operate under channel conditions that are characterized by a sparse channel impulse response. When the amount of training is given by the standard, compressed sensing channel estimation can exploit this sparsity to improve the quality of the channel estimate. In this paper, we analyze and compare the hardware complexity and denoising performance of three greedy algorithms for the 3GPP LTE system. The complexity/performance trade-off is analyzed using parameterized designs with varying configurations. One configuration of each algorithm is fabricated in a 180nm process and measured.

MP6-6 3:55 PM

A Low Energy High Speed Reed-Solomon Decoder Using Decomposed Inversionless Berlekamp-Massey Algorithm

Hazem A. Ahmed, Hamed Salah, Tallal ElShabrawy, German University in Cairo; Hossam A. H. Fahmy, Cairo university

This paper proposes an area efficient, low energy, high speed pipelined architecture for a Reed-Solomon decoder based on Decomposed Inversionless Berlekamp-Massey Algorithm, where the error locator and evaluator polynomial can be computed serially. In the proposed architecture, a new scheduling of t Finite Field Multipliers (FFMs) is used to calculate the error locator and evaluator polynomials to achieve a good balance between area, latency, and throughput. This architecture is tested in two different decoders. The first one is a pipelined two parallel decoder, as two parallel syndrome and two parallel Chien search are used. The second one is a conventional pipelined decoder, as conventional syndrome and Chien search are used. Both decoders have been implemented by 0.13

m CMOS IBM standard cells. The two parallel RS(255, 239) decoder has gate count of 37.6 K and area of 1.18 mm2, simulation results show this approach can work successfully at the data rate 7.4 Gbps and the power dissipation is 50 mW. The conventional RS(255, 239) decoder has gate count of 30.7 K and area of 0.99 mm2. Simulation results show this approach can work successfully at the data rate 4.85 Gbps and the power dissipation is 29.28 mW.

MP6-7 4:20 PM

Design of Large Polyphase Filters in the Quadratic Residue Number System

Gian Carlo Cardarilli, Università degli Studi di Roma “Tor Vergata”; Alberto Nannarelli, Technical University of Denmark; Yann Oster, Thales Alenia Space; Massimo Petricca, Marco Re, Università degli Studi di Roma “Tor Vergata”

In this work, we revisit the implementation of polyphase filter banks in Quadratic Residue Number System (QRNS) for banks with a large number of channels by developing a new design methodology suitable for large systems required in the new generation of systems on-board satellites. Furthermore, we compare the QRNS filter bank with an equivalent bank implemented in the traditional Complex Two’s Complement System (CTCS) in terms of throughput, area and power dissipation. The results for large filter banks confirm the earnings in power consumption by using the QRNS.

MP6-8 4:45 PM

FPGA Implementation Analysis of Polyphase Channelizer Performing Sample Rate Change Required for both Matched Filtering and Channel Frequency Spacing

Mehmod Awan, Aalborg University; fred harris, San Diego State University; Chris Dick, Xilinx, Inc.; Peter Koch, Aalborg University

The paper presents the architectural domain analysis for FPGA (Field Programmable Gate Array) implementation of a polyphase filter bank channelizer with an embedded square root shaping filter in its polyphase engine that performs two different re-sampling tasks required for spectral shaping and for M-channel channelizer. Algorithms: Radix-2 FFT, Prime Factor and Winograd Fourier Transform are considered for IFFT, where as polyphase filter is analyzed in terms of symmetric structure, serial polyphase structures with serial and parallel MAC approaches. The computational workload for these algorithms and their implementation structures are presented together with their hardware mapping to Virtex-5 FPGA by exploiting the inherent parallelism. Their resource utilization is presented along with different optimization techniques.

Track 8 – H. Speech, Image and Video Processing

Session: MPa7 – Video Compression

Chair: ***Cheolhong An,*** *Qualcomm, Inc.*

MP7a-1 1:30 PM

Spectral Entropy-Based Quantization Matrices for H.264/AVC Video Coding

Malavika Bhaskaranand, Jerry Gibson, University of California, Santa Barbara

In transform-based compression schemes, the task of choosing, quantizing, and coding the coefficients that best represent a signal is of prime importance. As a step in this direction, Yang and Gibson have designed a coefficient selection mechanism based on Campbell's coefficient rate and spectral entropy. Building on this mechanism, we develop a scheme to allocate bits amongst the chosen coefficients that can outperform the classical method under certain conditions. We then design quantization matrices (QMs) based on the proposed bit allocation method. Results show that the newly designed QMs perform better than the default QMs for H.264/AVC encoding in terms of both PSNR and structural similarity (SSIM). The proposed method entails delay but is not computationally intensive.

MP7a-2 1:55 PM

Motion Blur Adaptive Rate Control

Cheolhong An, Qualcomm Inc.

In this paper, a new Rate Control (RC) algorithm is proposed to improve perceptual quality using motion blur estimation. The blurriness of captured video frames is estimated from a global motion vector, encoding frame rate and exposure time. For given blurriness of frames, the proposed RC algorithm reallocates coding bits between blurry frames and sharp frames. From the experimental results, the proposed RC algorithm achieves much higher quality after a motion blur region and similar quality in a motion blur region comparing to the RC algorithm in the JM reference software model.

MP7a-3 2:20 PM

Compressive Sensing based Multiview Image coding with Belief Propagation

Parmida Beigi, Xiaoyu Xiu, Jie Liang, Simon Fraser University

In multiview video setup, acquisition cost is greatly reduced if some of the cameras can operate at lower quality. Compressed Sensing (CS) can be used for this purpose, that recovers signals from far fewer samples than conventional methods use. One of CS methods leading to a fast and efficient encoding/decoding process is Belief Propagation (BP). It uses low density parity check (LDPC) codes for the sensing, and message passing method for the decoding system. The proposed idea is to use such method in multiview video setting, by making use of view interpolation techniques for signal recovery. Numerical results show performance improvement of using side information (view interpolation) in the recent BP recovery system (Baron et al., 2010).

MP7a-4 2:45 PM

Frame Corruption Estimation from Route Messages for Video Coding over Mobile Ad Hoc Networks

Yiting Liao, Jerry Gibson, University of California, Santa Barbara

Recently we proposed a cross-layer design to support video communication over error-prone mobile ad-hoc networks. The idea is to utilize routing messages and network parameters to estimate the corrupted frames, and to guide the reference frame selection at the video encoder to mitigate error propagation. In this paper, we focus on the frame corruption estimation method used in the design. We build a packet loss probability model from the MAC layer mechanism and network parameters; then we utilize the model along with the routing messages received at the network layer to estimate the possible corrupted frames. We study our estimation method under different network settings and demonstrate its effectiveness and robustness. We further show the video quality gains achieved by adapting reference frame selection to the estimated results.

Session: MPa8 – Adaptive Signal Processing in Communications 1:30 PM – 3:10 PM

Chair: ***Martin Haardt,*** *Technical University Ilmenau*

MP8a4-1
A New Algorithm for Sidelobe Suppression and Performance Comparison in DFT-OFDM Cognitive Radios

Mohamed Marey, Octavia A. Dobre, Memorial university; Tricia Willink, Communications Research Center

In this paper, we propose a novel algorithm to reduce the out-of-band radiation for the Discrete Fourier Transform (DFT)-based OFDM in a CR environment. Further, we provide a performance comparison for the proposed algorithm and conventional algorithms in terms of power spectral density (PSD), peak to average power ratio (PAPR), and required bandwidth for overhead. Simulation results demonstrate that the proposed algorithm outperforms the conventional algorithms at the cost of an overhead, with a similar PAPR. The trade-off between the PSD (as a benefit) and the PAPR and overhead (as costs) is emphasized, being used for the selection of specific algorithm parameters.

MP8a4-2
An Iterative Widely Linear Interference Suppression Algorithm based on Auxiliary Vector Filtering

Lei Wang, University of York; Nuan Song, Ilmenau University of Technology; Rodrigo C. de Lamare, University of York; Martin Haardt, Ilmenau University of Technology

We introduce a new Widely Linearly (WL) framework that combines the WL filter with the Auxiliary Vector Filtering (AVF) technique and develop an adaptive algorithm for interference suppression in a high data rate Direct Sequence Ultra Wideband (DS-UWB) system. The proposed algorithm exploits the second-order behavior of the received signal and takes full advantage of the improper property nature of the non-circular data. It utilizes an iterative procedure to update the WL weight vector. The key properties of the proposed algorithm corresponding to the WL processor are analyzed. Simulation results are provided to show the superior performance of the proposed algorithm over its linear counterpart and conventional linear/WL Minimum Mean Square Error (MMSE) adaptive algorithms.

MP8a4-3
Non-negative Distributed Regression for Data Inference in Wireless Sensor Networks

Jie Chen, Université de Technologie de Troyes; Cédric Richard, Université de Nice Sophia-Antipolis; Paul Honeine, Université de Technologie de Troyes; Jose Carlos M. Bermudez, Federal University of Santa Catarina

Wireless sensor networks are designed to perform on inferences about the environment they are sensing. Due to the inherent physical characteristics of systems under investigation, non-negativity is a desired constraint that can be imposed on the system parameters in some real-life phenomena sensing tasks. In this paper, we propose a kernel-based machine learning strategy to deal with regression problems. Multiplicative update rules are derived in this context to ensure the non-negativity constraints to be satisfied. A distributed algorithm that requires only communication between neighbors is proposed to cope with typical limited energy and bandwidth resources. Synthetic data managed by heat diffusion equations are used to compare the proposed and known algorithms and to illustrate their tracking capabilities.

MP8a4-4
Blind Adaptive Equalizer Based on PDF Matching for Rayleigh Time-Varying Channels

Adel Daas, Stephan Weiss, University of Strathclyde

In this paper, we propose a new adaptive technique for blind equalisation for fast time-varying channels. The proposed approach is based on fitting the probability density function (pdf) of the equalizer output to the desired pdf of the corresponding symbol alphabet. The underlying pdf at the equalizer output is estimated by means of the Parzen Window method. The cost function of the proposed technique can be measured by a stochastic gradient descent approach. The performance of the proposed adaptation strategy is assessed by a number of simulations, and benchmarked against CMA under QPSK modulation in a doubly-dispersive environment. The channel has been implemented by producing a vector of Rayleigh faded coefficients which are drawn from complex Gaussian distribution.

MP8a4-5
A Systematic Approach to Incorporate Deterministic Prior Knowledge in Broadband Adaptive MIMO Systems
Herbert Buchner, Berlin University of Technology

Various approaches for incorporating prior system knowledge into adaptive filtering algorithms exist, e.g., using constrained adaptation. Moreover, also the basic setup of the adaptation problem, e.g., whether it is supervised or blind, can be considered as prior system knowledge. In this paper, we consider a systematic approach to incorporate such deterministic prior knowledge in broadband adaptive MIMO systems by optimizing the coefficients in arbitrary partly smooth manifolds. The resulting generic set of update equations explicitly shows all the available degrees of freedom for a top-down algorithm design. Using practically relevant examples, we show how both well-known and novel algorithms for various applications can be derived using the framework.

Track 1 – A. Communications Systems

Session: MPa8 – Communication Systems II 1:30 PM – 3:10 PM

Chair: ***Martin Haardt,*** *Technical University Ilmenau*

MP8a1-1
Uplink Interference Scenarios in Two-Tier Networks
Zhenning Shi, Alcatel Lucent - Shanghai Bell; Mark Reed, Ming Zhao, National ICT Australia (NICTA); He Wang, Australian National University

In this paper, a two-tier UMTS network is considered where a large number of randomly deployed Wideband Code Division Multiple Access (WCDMA) femtocells are laid under macrocells where the spectrum is shared. The co-channel interference between the cells may be a potential limiting factor for the system. We study the uplink of this hybrid network, and identify the critical scenarios that give rise to substantial interference. The mechanism for generating the interference is analyzed and guidelines for interference mitigation are provided. The impacts of the cross-tier interference, specially caused by increased numbers of users and higher data rates is evaluated in the multi-cell simulation environment in terms of the noise rise at the base stations, the cell throughput and the user transmit power consumption.

MP8a1-2
Rethinking Capacity Per Unit Cost
Matthew Nokleby, Behnaam Aazhang, Rice University

Verdu's seminal work offers an elegant characterization of capacity per unit cost, but only when a cost-free input symbol is available. Furthermore, optimal transmission strategies under such scenarios result in infinitesimal transmission rates. We argue that the existence of a cost-free symbol is unrealistic and that the resulting low-rate transmission strategies are not useful in practice. We therefore examine capacity per cost when no cost-free symbol is available. In this case, the corresponding transmission rates are bounded away from zero. Specifically we propose a cost function for wireless channels that balances both energy and temporal costs. Using this cost function, we explore the capacity per cost regions of single- and multiple-user channels.

MP8a1-3
Efficiency of Rate-maximization Game Under Bounded Channel Uncertainty
Amod J.G. Anandkumar, Advanced Signal Processing Group; Animashree Anandkumar, Laboratory for Information and Decision Systems; Sangarapillai Lambotharan, Jonathon Chambers, Advanced Signal Processing Group

The problem of competitive rate maximization is an important signal-processing problem for power-constrained multi-user systems. It involves solving the power control problem for mutually interfering users operating across multiple frequencies. The previously introduced robust rate-maximization approach helps find the solution in systems with bounded channel uncertainty. Here, we analyse the effect of uncertainty on the global efficiency of the robust rate-maximization game. We show that the robust equilibrium tends to move towards FDMA solution as uncertainty increases and thus increases the sum-rate for systems where FDMA is Pareto-optimal and verify the results through simulations.

MP8a1-4

A Semidefinite Programming Approach to Cooperative Localization in Wireless Sensor Networks

Ning Wang, Liuqing Yang, University of Florida

In this paper, we propose a novel semidefinite programming (SDP) approach to the cooperative localization in wireless sensor networks. Compared with the noncooperative localization, the cooperative technique can improve the accuracy and enlarge the coverage of the localization system. However, exploiting the cooperation among the location-unaware sensor nodes will result in a higher computational complexity which has a large impact on the speed of localization. To solve this problem, we adopt the SDP technique to reduce the computational complexity. Our approach applies to both the Time-of-Arrival (ToA) and the Received Signal Strength (RSS) model. Simulations show that the cooperative localization with SDP can achieve good performance with considerably reduced complexity.

MP8a1-5

A Self-Organizing Solution for Interference Avoidance in TDD Underlay Femtocells

Francesco Pantisano, Centre for Wireless Communication (CWC) + Dipartimento di Elettronica Informatica e Sistemistica (DEIS); Kaveh Ghaboosi, Mehdi Bennis, Centre for Wireless Communication (CWC); Roberto Verdone, Dipartimento di Elettronica Informatica e Sistemistica (DEIS)

In this paper, we draw our attention to the problem of interference avoidance in a macrocell-femtocell network. Further, macro user equipments (MUEs) operate in a Frequency Division Duplexing (FDD) mode while all femtocells employ the macrocellular uplink spectrum in a Time Division Duplexing (TDD) mode. We propose a self-adapting algorithm to tackle highly interfering MUEs through handover procedure. Preliminary results reveal that the proposed solution provides lower outage probability than closed and open access policies, without significant loss in terms of signal-to-noise-and-interference ratio.

MP8a1-6

GLRT Based Cooperative Spectrum Sensing with Location Information

Ning Han, Hongbin Li, Jun Fang, Stevens Institute of Technology

In this paper, we address the problem of cooperatively detecting a primary user with unknown transmit power among multiple cognitive radio (CR) users with their location information available at the CR base station. By assuming the noise variances at each CR user are equal and known, the reporting process is lossless, a generalized likelihood ratio test (GLRT) is developed at the CR base station to first estimate the transmit power of primary user and then detect the primary user signal. Performance comparison between the cooperative GLRT and the OR-Rule based method is provided. Meanwhile, asymptotic performance analysis for the proposed GLRT is also presented. Simulation results show that the proposed detector exhibits better performance than conventional techniques in terms of detection probability.

MP8a1-7

Iterative Decoding on Divided Trellis for Turbo Codes

Jinhong Wu, Atheros Communications, Inc.; Branimir Vojcic, Jia Sheng, George Washington University

A simple alternative to the sliding window algorithms for turbo decoding is presented. It avoids the use of guard windows. The trellis associated with each constituent code is divided into non-overlapping short segments and iterative decoding is performed within each segment. After each iteration, the state probabilities of each segment's boundaries are stored and used for the next iteration. The fast convergence of the stored soft metrics allows for the decoding to be as effective as that by standard decoding process without using any initial recursions.

MP8a1-8

Detection of CPM Based on Second-Order Cyclostationarity

Amy Malady, A. A. (Louis) Beex, Virginia Tech

For complex CPM signals, depending on the size of the symbol alphabet and the values and number of modulation indices, evaluation of cyclostationarity at various high orders may be required. We present a novel detection method based on the cyclostationarity present in the individual real and imaginary components of CPM signals. Second-order cyclostationarity exists for certain delay vectors in 1REC CPM signals, regardless of the size of the symbol alphabet or the values and number of modulation indices. This low-order cyclostationarity is exploited to perform blind, asynchronous detection of CPM signals in the presence of complex additive Gaussian noise.

MP8a1-9

Determination of Cyclic Delay for CDD Utilizing RMS Delay Spread in OFDMA Multiuser Scheduling Systems

Seong-Ho Hur, University of California, San Diego; Min-Joong Rim, Dong-Kook University; Bhaskar D. Rao, James R. Zeidler, University of California, San Diego

The multiuser diversity gain increases with the degree and dynamic range of channel fluctuations. For a multi-antenna OFDM system operating in a channel with limited fluctuations, cyclic delay diversity (CDD) was proposed to increase frequency selectivity. This is achieved by introducing a cyclic delay between the different transmit antennas thereby increasing the effective number of paths in the resulting channel. In this paper, for a multiuser system, we propose a scheme to determine per-user optimal cyclic delay based on the knowledge of the RMS delay spread of the channel. We show that the proposed technique achieves better performance than a conventional fixed cyclic delay scheme and that the throughput is very close to the optimal sum rate possible with CDD.

MP8a1-10

Effective SINR Distribution in MIMO OFDM Systems

Alexandra Oborina, Visa Koivunen, Helsinki University of Technology; Tero Henttonen, Nokia Oyj

In wireless networks, the system level performance after channel decoding is accurately evaluated by means of packet error rate. It is used to develop the link adaptation algorithms, e.g. packet scheduling, hybrid-ARQ, adaptive modulation and coding, etc. In case of multi-stream multi-carrier wireless transmission that provides high data rates required by future beyond 3G and 4G wireless networks, the effective SINR in combination with AWGN curves is used to determine the packet error rate. The effective SINR is a commonly used channel quality measure obtained through an one-dimensional mapping of K post-processed subcarrier SINR values and a scaling parameter defined for each modulation and coding scheme. In this paper we analyze the distribution of the effective SINR taking into account the correlations properties of the subcarrier SINR values after MMSE receiver.

MP8a1-11

Pilot Design for OFDM-Based Non-Regenerative Relay Networks in the Presence of Phase Noise

Payam Rabiei, Won Namgoong, Naofal Al-Dhahir, University of Texas at Dallas

The bit error rate performance of orthogonal frequency division multiplexing (OFDM) systems is severely degraded in the presence of phase noise. This performance degradation is more pronounced over non-regenerative (amplify-and-forward) relay networks due to an excessive interference from the relay node. In this paper, a pilot structure is proposed with a certain set of design parameters which are computed to minimize the interference at the destination node. These parameters are specifically optimized for an OFDM-based non-regenerative relay network in the presence of phase noise.

MP8a1-12

Sum-Rate Maximization by Bandwidth Re-allocation for Two Users in Partial Frequency Reuse Cellular Networks

Bujar Krasniqi, Technische Universität Wien; Martin Wolkerstorfer, FTW Forschungszentrum Telekommunikation Wien; Christian Mehlfuehrer, Christoph Mecklenbrauker, Technische Universität Wien

In this paper we apply constrained optimization techniques to optimally allocate bandwidth and transmit power to the users in a cellular network. We utilize partial frequency reuse with multiple users in the full and partial frequency regions as inter-cell interference mitigation technique. We show that the non-convex sum-rate maximization problem becomes convex under some simplifying assumptions. Moreover, an efficient and problem specific algorithm is developed to solve the problem for a fixed bandwidth allocation. Our results show that in the optimum, the full amount of power is assigned to the users. We further demonstrate that re-allocating the outer bandwidth to the inner users when no outer users are present, results in an increased sum-rate.

MP8a1-13
Performance of UWB MIMO Relay Systems in Real UWB Channels
Kiattisak Maichalernnukul, Trung Kien Nguyen, Feng Zheng, Thomas Kaiser, University of Hannover

In this paper, the performance of various ultrawideband (UWB) multiple-input multiple-output (MIMO) relay systems is evaluated for the first time based on UWB channel measurements. In particular, the measurement results for UWB relay channels in line-of-sight as well as non-line-of-sight indoor environments are presented. The average bit error rates of dual-hop MIMO relay systems and cooperative MIMO relay systems are examined and compared in the measured channels.

Track 8 – H. Speech, Image and Video Processing
Session: MPa8 – Speech Enhancement 1:30 PM – 3:10 PM
Chair: ***David Anderson,*** *Georgia Institute of Technology*

MP8a2-1
Combined Reduction of Time Varying Harmonic and Stationary Noise Using Frequency Warping
Thomas Esch, Matthias Rüngeler, Florian Heese, Peter Vary, RWTH Aachen University

Speech enhancement under non-stationary environments is still a challenging problem. This contribution presents a noise reduction system that is capable of tracking and suppressing both time varying harmonic noise and stationary noise. In a first step, the harmonic noise power is estimated using a modified Minimum Statistics approach that performs frequency warping according to the harmonic's fundamental frequency. A conventional noise estimation technique is applied in a second step in order to reduce the random components of the noise spectrum. The performance of the proposed noise suppression system is shown to be consistently better than conventional approaches.

MP8a2-2
Comparison of Various Adaptive Kalman Filtering Algorithms Applied to Single Microphone Blind Audio Source Separation
Siouar Bensaid, Dirk Slock, Eurecom

In this paper, we compare the adaptive EM-Kalman algorithm to an improved version of the Extended Kalman algorithm (EKF) within the context of single microphone blind source separation. A linear state space model with unknown parameters is derived. The separation is achieved by estimating the state as well as the unknown parameters of the state space model.

MP8a2-3
A MAP Criterion for Detecting the Number of Speakers at Frame Level in Model-based Single-Channel Speech Separation
Pejman Mowlaee, Mads Græsbøll Christensen, Zheng-Hua Tan, Søren Holdt Jensen, Aalborg University

The problem of detecting the number of speakers for a particular segment occurs in many different speech applications. In single channel speech separation, for example, this information is often used to simplify the separation process, as the signal has to be treated differently depending on the number of speakers. Inspired by the asymptotic maximum a posteriori rule proposed for model selection, we pose the problem as a model selection problem. More specifically, we derive a multiple hypotheses test for determining the number of speakers at a frame level in an observed signal based on underlying parametric speaker models, trained a priori. The experimental results indicate that the suggested method improves the quality of the separated signals in a single-channel speech separation scenario at different signal-to-signal ratio levels.

MP8a2-4
Toward Overcoming Fundamental Limitation in Frequency-Domain Blind Source Separation for Reverberant Speech Mixtures
Lae-Hoon Kim, Mark Hasegawa-Johnson, University of Illinois at Urbana-Champaign

Blind source separation can be implemented in the frequency domain using a one-tap multiplication operation in each frequency bin, but only if the transform window is long enough to disregard temporal aliasing effects. If we take a short-time frequency transformation with a window shorter than the room reverberation time, the justification above does not hold anymore. We present an appropriate representation in the short-time frequency domain. The suitability is justified by showing the equivalence with the original time domain approach. Theoretical proof under the overlap-add context is provided, and experimental validation using a corpus synthesized by convolution with a measured impulse response.

MP8a2-5
Auditory Motivated Analysis Based Speech enhancement
Novlene Zoghlami, Zied Lachiri, ENIT

We propose a new speech enhancement method based on auditory properties of human ear using non-uniform and multi-band analysis. The noisy signal is divided into a number of sub-bands using a gammachirp filter bank with non-linear ERB resolution, the ERB sub-signals are then individually manipulated according the Log spectral magnitude MMSE estimator to obtain the enhanced speech signal. Performance evaluation demonstrates significant improvements results over classical speech enhancement algorithm, when tested with speech signals corrupted by real world noise conditions.

MP8a2-6
Modified Fermat Transforms for Reliable and Efficient De-noising of Speech Signals
Chandra Radhakrishnan, Kenneth Jenkins, Pennsylvania State University; Carnell Hunter, Virginia Commonwealth University; Robert Nickel, Bucknell University

Recently the Modified Fermat Number Transform (MFNT) based on Right Circular Convolution (RCC) was extended to form a Quadratic MFNT (QMFNT) by introducing Left Circular Convolution (LCC) and interpreting the combined result as a quadratic representation of the resulting convolution output. The QMFNT enables efficient convolution/correlation to be implemented by overlap-add block processing without zero padding, resulting in improved computational efficiency and potentially reduced power requirements for nanoscale VLSI implementations. This paper investigates the use of an efficient QMFNT algorithm that is capable of implementing a novel speech de-noising algorithm that has potential applications in speech enhancement and speech recognition.

Track 8 – H. Speech, Image and Video Processing

Session: MPa8 – Selected Topics in Speech and Audio 1:30 PM – 3:10 PM

Chair: ***Jerry Gibson,*** *University of California, Santa Barbara*

MP8a3-1
Frequency Dependent GTD Coders
Ching-Chih Weng, P. P. Vaidyanathan, California Institute of Technology

This paper proposes the frequency dependent generalized triangular decomposition (FDGTD) coder family for wide-sense-stationary (WSS) vector processes. Under the uniform bit allocation constraint, a set of necessary and sufficient conditions for FDGTD's coding gain optimality is derived. It is shown that one member in the FDGTD family, the frequency dependent geometric mean decomposition (FDGMD) coder, satisfies these conditions and thus is optimal. It is also demonstrated that the FDGMD coders use a simpler uniform quantizer structure and yet achieve a better performance than the conventional optimal orthonormal subband coders with sophisticated bit allocation scheme.

MP8a3-2
Time-Scale Modification of Audio Signals Using Multi-Relative Onset Time Estimations in Sinusoidal Transform Coding
Jonathan Kim, Mark Clements, Georgia Institute of Technology

The traditional sinusoidal transform coder (STC) was originally developed for analysis, synthesis, and modification of speech and other non-polyphonic audio signals. In this paper, a novel method of time-scale modification for polyphonic and multi-pitch audio signals based on STC is proposed. The proposed method does not require a pitch estimation and as such it enables STC based algorithms to perform modifications on polyphonic audio signals. The frequency jitter artifacts in the traditional STC are mostly due to the inaccurate onset time estimates measured by pitch periods. The proposed method eliminates the frequency jitter artifacts significantly by using multi-onset time estimations.

MP8a3-3

Tandeming Analysis of Perceptual Pre-weighting and Post-weighting Multimode Tree Coder

Ying-Yi Li, Pravin Ramadas, Jerry Gibson, University of California, Santa Barbara

The perceptual pre-weighting and post-weighting Multimode Tree Coder is low delay and low complexity. Since the tandem connection of different codecs in voice calls is common today, it is also important to assess any loss in end-to-end speech quality caused by asynchronous tandem coding. We evaluate the tandeming performance of our Multimode Tree Coder when tandemed with itself, with G.727, and with the AMR-NB codec. The results show that the tandem performance of the Multimode Tree Coder is comparable to the AMR-NB coder at 12.2 kbps.

MP8a3-4

Improved Approach for Calculating Model Parameters in Speaker Recognition Using Gaussian Mixture Models

Prashant Metkar, Aaron Cohen, Keshab Parhi, University of Minnesota-Twin Cities

In speaker identification, most of the computation is due to the distance or likelihood calculation between feature vectors of the test signal and the speaker model in the database. The time required for identifying a speaker is a function of feature vectors and their dimensionality and the number of speakers in the database. In this paper, we focus on optimizing the performance of Gaussian mixture (GMM) based speaker identification system. An improved approach for model parameter calculation is presented. The advantage of proposed approach lies in the reduction in computational time by a significant amount over an approach which uses expectation maximization (EM) algorithm to calculate the model parameter values. This approach is based on forming clusters and assigning weights to them depending upon the number of mixtures used for modeling the speaker. The reduction in computation time depends upon how many mixtures are used for training the speaker model.

MP8a3-5

An Efficient Constant-Q Spectrum Analyzer Architecture Using Polyphase Filter Bank

Xiaofei Chen, San Diego State University; Elettra Venosa, Seconda Università degli Studi di Napoli; fred harris, San Diego State University

In this paper we present an extremely efficient constant-Q spectrum analyzer. By using half-band filtering and 2-to-1 down sampling, the traditional analyzer successively shifts lower octaves to the same spectral interval so that the same post-processing can be applied to all of them. To decrease the total workload of such a structure a 4-path polyphase filter could be used to perform a 4-to-1 down sampling before processing the octaves in the proportional bandwidth filter bank; this engine down samples the current octave simultaneously translating it from quarter sample rate to baseband. In this paper we do much better! We propose a spectrum analyzer that uses only a 4-path polyphase channelizer along with a proportional bandwidth filter bank. The 4-path polyphase channelizer is used to perform 4-to-2 down-sampling. It is able to simultaneously shift lower octaves to the same spectral interval and translate the current octave to base-band allowing the proportional filter bank to work on a reduced sampling rate and decreasing the total workload of the system. To further reduce the workload recursive filters are used to implement the proportional bandwidth filters. A workload analysis along with a comparison between traditional and proposed constant-Q spectrum analyzer is also shown in this paper.

MP8a3-6

Localization Based on Source Sparsity

Petros Boufounos, Mitsubishi Electric Research Labs; Bhiksha Raj, Carnegie Mellon University

Recent work has demonstrated the power of sparse models and representations in signal processing applications and has provided the community with computational tools to use it. In this paper we explore the use of sparsity in localization and beamforming when capturing multiple audio sources using a microphone array. Specifically, we reformulate the wideband signal acquisition as a block sparsity problem, in a combined frequency-space domain. Signals from a spatially sparse set of sources have support in all frequencies. Using techniques from the model-based compressive sensing literature we demonstrate that it is possible to robustly capture, reconstruct and localize multiple signals present in the scene.

MP8a3-7

Low Complexity 3D Source Localization Using Pseudointensity Vectors

Daniel P. Jarrett, Emanuël A.P. Habets, Patrick A. Naylor, Imperial College London

While linear microphone arrays have been widely studied throughout the last few decades, spherical microphone arrays have also more recently been the focus of much research due to their ability to analyse sound fields in three dimensions [1, 2, 3]. In [4] we proposed a solution to the problem of localizing a single static acoustic source, using a spherical harmonic domain method based

on a pseudointensity vector. In this paper we extend this method to multiple and/or moving sources. When compared to other localization methods (e.g., energy based), our method offers substantially reduced computational complexity, while retaining good accuracy even in a reverberant environment.

MP8a3-8
Cyclic Matching Pursuits with Multiscale Time-Frequency Dictionaries
Bob Sturm, Mads Græsbøll Christensen, Aalborg University

We investigate the performance of cyclic matching pursuit (CMP) to the sparse approximation of signals with time-localized phenomena over multiscale time-frequency dictionaries. CMP was proposed and studied within the context of parametric perceptual audio coding with a dictionary of frequency-localized Fourier functions. Furthermore, we propose an orthogonal extension of CMP. Overall, we find that the cyclic approach of CMP produces signal models that are competitive with existing greedy iterative descent methods, such as matching pursuit (MP), orthogonal MP (OMP), and orthogonal least squares (OLS), with respect to the Euclidean norm of approximation error.

Track 5 – E. Array Processing and Statistical Signal Processing
Session: MPa8 – Array-based Estimation 1:30 PM – 3:10 PM
Chair: ***John Shynk,*** *University of California, Santa Barbara*

MP8a5-1
Emitter Position and Velocity Estimation Given Time and Frequency Differences of Arrival
Alon Amar, Geert Leus, Delft University of Technology; Benjamin Friedlander, University of California, Santa Cruz

Consider the problem of estimating the position and the velocity of an emitter given time and frequency differences of arrival acquired by a passive sensor array. By jointly eliminating the non-linear nuisance parameters of the model using an appropriate orthogonal projection, we obtain a least squares estimator of the parameters of interest. We show that the estimator is asymptotically unbiased and express its covariance matrix. Simulations show that the performance of the method is similar to a previously suggested two-step estimator with less complexity, and close to the \CRLB at high signal to noise ratio.

MP8a5-2
Exploring Sensitivity of Joint Diagonalization in Convolutive Blind Source Separation
Savaskan Bulek, Nurgun Erdol, Florida Atlantic University

This paper investigates several factors affecting the sensitivity of joint approximate diagonalization of a set of time varying cross-spectral matrices for blind separation of convolutive mixtures of speech signals. We study the effect of number of matrices in this set, and show that estimation of demixing system parameters is related to both several statistics of the perturbation term, occuring due to nonvanishing cross-spectra, and uniqueness of the joint diagonalizer measured by modulus of uniqueness parameter. Moreover, the effects of the number of multiple windows, to be used in cross-spectrum estimation, on the separation performance are examined via numerical simulations.

MP8a5-3
Blind Phase-Shift-Based DOA Estimation
John Shynk, Sheng-Luen Wei, University of California, Santa Barbara

Blind direction-of-arrival (DOA) estimation algorithms based on direction vector phase shifts in a uniform linear array are investigated in this paper. Instead of using a pilot signal for training, the output of a conventional beamformer is used to compute the phase difference relative to the output of a second beamformer. This is possible because the two sets of beamformer weights are related to each other in a specific and known manner: (i) they are identical for a subarray approach and (ii) conjugated for a reversed beamformer. Using a least-squares (LS) model, these techniques are compared with another blind DOA estimator based on the the constant modulus (CM) array. Computer simulations are provided to verify the analytical results and to evaluate the performance of each approach for various cochannel interference and noise conditions.

MP8a5-4
A Joint AOA, AOD and Delays Estimation of Multipath Signals based on Beamforming Techniques
Ismehene Chahbi, Badii Jouaber, Institut TELECOM, Telecom SudParis

Angle of Departure (AOD) and Delay of Arrival (DOA) information is added to beamforming techniques to provide improved and more robust estimates of Angle of Arrival (AOA) and localization. A joint AOA, AOD and DOA algorithm based on the Capon Beamformer is proposed to reduce complexity and computation time compared to subspace-based methods. The proposed approach works even if the number of multipaths exceeds the number of antenna elements. Simulation results are provided to assess the improvements in channel estimation for a multiple antenna array.

MP8a5-5
Using Moment Finite Rate of Innovation for LIDAR Waveform Complexity Estimation
Juan Castorena, Charles Creusere, David Voelz, New Mexico State University

LIDAR sensing collects data to obtain detailed topographical information of a region. A major challenge in using such technologies is the large amount of data needs to be collected for accurate surface reconstruction which in turn imposes significant storage, processing and transmission requirements. Current efforts to solve this problem have been focusing on designing compression algorithms to lower the sampling rate requirements. However, these require the collection of large amounts of data, most of which is ultimately discarded by the processing. Instead, in this study our approach to compression is to model individual laser return pulses acquired at spatial sampling rates lower than the traditional Nyquist/Shannon limit by using sampling moments and reconstructing signals of finite rate of innovations. Our results show that accurate classifications of waveform complexity with the simple nearest neighbor classifier are obtained using the proposed algorithm.

MP8a5-6
Hybrid Tensor Decomposition for Sound Source Separation
Na Li, Carmeliza Navasca, Clarkson University

We present a method for separating signals from a mixture of several sound sources in multi-channel. Based on the nonnegative matrix factorization and the sparsity of power spectrogram of signals, the hybrid tensor decomposition combines the ℓ_1 minimization to obtain sparse representation of signals and the nonnegative matrix factorization for factors containing amplitude basis or source gains. Our algorithm is tested to separate the instrument sources from a two channel mixtures containing a clarinet, a piano and a steel drum.

MP8a5-7
Multi-Objective Optimized OFDM Radar Waveform for Target Detection in Multipath Scenarios
Satyabrata Sen, Arye Nehorai, Washington University in St. Louis

We propose a multi-objective optimization technique to design an orthogonal frequency division multiplexing (OFDM) radar signal for detecting a static target in the presence of multipath reflections. We employ an OFDM signal to increase the frequency diversity of the system as different scattering centers of a target resonate variably at different frequencies. Moreover, the multipath propagation increase the spatial diversity by providing extra “looks” at the target. First, we develop a parametric measurement model by reformulating the target detection problem as a sparse estimation method. At a particular range cell, we exploit the sparsity of multiple paths, be they line-of-sight (LOS) or reflected paths, and the knowledge of the environment to estimate along which path the target responses are received. Then, we evaluate the performance characteristics of the underlying detection test and show that the performance improves with the non-centrality parameter. On the other hand, to obtain an efficient sparse recovery, we need to have small value of the coherence measure. Based on these analyses, we formulate a constrained multi-objective optimization algorithm, to design the spectral parameters of the OFDM waveform, with two simultaneous objectives: maximize the non-centrality parameter and minimize the block-coherence measure. We present numerical examples to demonstrate the performance improvement due to adaptive waveform design.

Track 1 – A. Communications Systems

Session: MPb1 – Trends for Future Wireless Systems

Chair: ***Tom Marzetta,*** *Bell Labs*

MP1b-1 3:30 PM

Fading Models and Metrics for Contemporary Wireless Systems

Nihar Jindal, University of Minnesota; Angel Lozano, UPF

This paper examines some of the settings commonly used to represent fading. We raise the question of whether these settings remain meaningful in light of the advances that wireless communication systems have undergone over the last decade. A number of weaknesses are pointed out, and ideas on possible fixes are put forth. Some of the identified weaknesses have to do with models that, over time, have become grossly inadequate; other weaknesses have to do with changes in the operating conditions of modern systems, and others with the coarse and asymptotic nature of some of the most popular performance metrics.

MP1b-2 3:55 PM

Doubling Throughput in Cellular Networks with Higher-order Sectorization

Howard Huang, Reinaldo Valenzuela, Cuong Tran, Susan Walker, Dragan Samardzija, Bell Laboratories, Alcatel-Lucent

Wireless data usage in cellular networks is projected to increase an order of magnitude in the coming years. To address this coming tidal wave of data demand, conventional networks deploy three sectors per site and use multiple antenna (MIMO) techniques to achieve higher throughput. In contrast, we propose partitioning the site radially into twelve sectors per site and using single-antenna transmission per site. Higher-order sectorization has the advantage of being largely standards independent, simple to implement for both FDD and TDD systems, and compatible with single-antenna mobiles. For a given number of RF signals per site, it achieves similar performance compared to the capacity-achieving multiuser MIMO technique but with significantly lower complexity.

MP1b-3 4:20 PM

Performance of TDD-based MU-MIMO Systems: Multiuser Diversity Interference Mitigation and CSI Costs

Haralabos Papadopoulos, DOCOMO USA Labs; Giuseppe Caire, University of Southern California; Sean Ramprashad, DOCOMO USA Labs

It is now understood that the performance of schemes such as MU-MIMO and Network MIMO must take into account system overheads, including those for acquiring CSI. This paper considers TDD-based CSI acquisition schemes. Unlike FDD-based schemes, TDD-based training overheads do not grow with the number of base-station antennas. However, TDD overheads do scale with the size of the user-scheduling set per cell and with the cell pilot-reuse factor. In the regime of moderate numbers of base-station antennas, this yields interesting design tradeoffs balancing multi-user diversity and interference mitigation across cells. We consider a number of such schemes, assessing edge-user and cell-throughput performance under equitable fairness criteria.

MP1b-4 4:45 PM

Making MIMO Really Work: The 400-Antenna Base Station

Thomas Marzetta, Alexei Ashikhmin, Bell Laboratories, Alcatel-Lucent

Large-scale base station antenna arrays, time-division duplex (TDD) operation, and multi-user MIMO constitute a potent combination which potentially provides order-of-magnitude improvements over LTE Advanced with respect to throughput and reduced transmitted power, for everybody, irrespective of propagation conditions or mobility, and realizable with cheap dumb single-antenna terminals. A large excess of base station antennas over terminals is always beneficial: an ever-increasing number of antennas eliminates the effects of channel estimation errors, fast fading, and uncorrelated noise. In the limit of an infinite number of antennas the only remaining impairment is inter-cell interference which results from the re-use of reverse-link pilots.

Session: MPb2 – MIMO Relays

Chair: ***Ozgur Oyman,*** *Intel*

MP2b-1 3:30 PM

Beamforming for Network-coded MIMO Two-way Relaying

Taemin Kim, Bernd Bandemer, Arogyaswami Paulraj, Stanford University

We investigate transmit beamforming for the multiple-access phase of the network-coded MIMO two-way relaying. Since global optimization of the beamforming vectors is intractable or involves complex iterative algorithm, we propose a practical solution to opportunistically combine candidate precoders based on the individual channels. Each candidate is chosen to meet a specific subspace constraint to provide desirable constellation shaping for network coding at the relay. We present an efficient algorithm to find the precoder which is optimal within each subspace constraint. Numerical simulation shows that the proposed scheme approaches an ideal performance bound, providing an efficient and near-optimal solution to the beamforming problem.

MP2b-2 3:55 PM

Residual Self-Interference in Full-duplex MIMO Relays After Null-Space Projection and Cancellation

Taneli Riihonen, Stefan Werner, Risto Wichman, Aalto University

The practical implementations of full-duplex MIMO relays require efficient mitigation of the relay self-interference signal propagating from the transmit antenna array to the receive antenna array. We concentrate on two particular mitigation schemes: time-domain cancellation and spatial-domain suppression by null-space projection. Earlier literature shows that both of these schemes can eliminate the self-interference in idealistic setups. Our new contribution is to analytically evaluate the isolation obtained in practice by performing the mitigation in realistic scenarios where the performance is limited due to imperfect side information. As a new mitigation scheme, we also show how the combination of time-domain cancellation and null-space projection further reduces residual self-interference.

MP2b-3 4:20 PM

Self-Interference Suppression in Full-Duplex MIMO Relays

Panagiota Lioliou, Mats Viberg, Chalmers University of Technology; Mikael Coldrey, Fredrik Athley, Ericsson AB

Full-duplex relays can provide cost-effective coverage extension and throughput enhancement. However, the main limiting factor is the resulting self-interference signal which deteriorates the relay performance. In this paper, we propose a novel technique for self-interference suppression in full-duplex Multiple Input Multiple Output (MIMO) relays. The relay employs transmit and receive weight filters for suppressing the self-interference signal. Unlike existing techniques that are based on zero forcing of self-interference, we aim at maximizing the ratio between the power of the useful signal to the self-interference power at the relay reception and transmission. Our simulation results show that the proposed algorithm outperforms the existing schemes since it can suppress interference substantially with less impact on the useful signal.

MP2b-4 4:45 PM

Optimal Channel Estimation and Training Design for MIMO Relays

Ting Kong, Yingbo Hua, University of California, Riverside

This paper considers a channel estimation scheme for a two-hop non-regenerative MIMO relay system. The scheme has two phases. In the first phase, the source node transmits no signal, the relay transmits a training matrix and the destination estimates the relay-to-destination channel matrix. In the second phase, the source transmits a training matrix, the relay amplifies and forwards with a relay transformation matrix, and the destination estimates the source-to-relay channel matrix. The estimation in each phase is done by the linear minimum mean squared error (LMMSE) method. The correlation in each channel matrix is modeled as a Kronecker product of transmit and receive correlations. Fast algorithms for computing the optimal training matrices at the source and the relay in both phases are developed by using the convex optimization and majorization theory.

Track 3 – C. Networks

Session: MPb3 – Learning and Optimization in Dynamic Networks

Co-Chairs: ***Qing Zhao,*** *University of California, Davis* ***and Keqin Liu,*** *University of California, Davis*

MP3b-1 3:30 PM

Distributed Learning Under Imperfect Sensing in Cognitive Radio Networks

Keqin Liu, Qing Zhao, University of California, Davis; Bhaskar Krishnamachari, University of Southern California

We consider a cognitive radio network, where M distributed secondary users search for spectrum opportunities among N independent channels without information exchange. The occupancy of each channel by the primary network is modeled as Bernoulli process with unknown mean which represents the unknown traffic load of the primary network. In each slot, a secondary transmitter chooses one channel to sense and subsequently transmit if the channel is sensed as idle. Sensing is considered to be imperfect, i.e., an idle channel can be sensed as busy and vice versa. Users transmit on the same channel collide and none of them can transmit successfully. The objective is to maximize the system throughput under the collision constraint imposed by the primary network while ensuring synchronous channel selection between each secondary transmitter and its receiver. The performance of a channel selection policy is measured by the system regret, defined as the expected total performance loss with respect to the optimal performance under the ideal scenario where all channel means are known to all users and collisions among users are eliminated throughput perfect scheduling. We show that the optimal system regret rate is at the same logarithmic order as the centralized counterpart with perfect sensing. An order-optimal decentralized policy is constructed to achieve the logarithmic order of the system regret rate while ensuring the fairness among all users.

MP3b-2 3:55 PM

The Asymptotics of Duplication-Deletion Random Graphs

Maziyar Hamdi, Vikram Krishnamurthy, University of British Columbia

This paper considers the dynamics of a duplication-deletion graph where at each time instant, one node can either join or leave the network. We derive the power law coefficient for this graph together with probability distribution of expected degrees. We show via a functional central limit theorem that the asymptotic scaled difference between the sample path and expected degree converges weakly to a Brownian bridge process. By exploiting the special structure of the adjacency matrix in the duplication-deletion model, a recursive bound is derived for the largest singular value of the adjacency matrix which can be used for dimension-reduction for large random graphs.

MP3b-3 4:20 PM

No-Regret Routing for Ad-hoc Wireless Networks

Abhijeet Bhorkar, Tara Javidi, University of California, San Diego

We consider the problem of optimal adaptive routing in wireless ad-hoc networks with respect to the regret (or learning loss) criterion. The regret criterion accounts for the loss in performance incurred because of the implicit on-line learning task involved when the system model is not completely known. In particular, we seek the construction of asymptotically efficient adaptive control schemes (i.e., schemes that minimize the rate at which regret accumulates with time), and/or adaptive schemes that come arbitrarily close to the minimum rate. The intention is to capture important issues in the conflict between learning and control.

MP3b-4 4:45 PM

Dynamic Optimization and Learning for Renewal Systems

Michael Neely, University of Southern California

We consider the problem of optimizing time averages in systems with independent and identically distributed behavior over renewal frames. This includes scheduling and task processing to maximize utility in stochastic networks with variable length scheduling modes. Every frame, a new policy is implemented that affects the frame size and that creates a vector of attributes. An algorithm is developed for choosing policies on each frame in order to maximize a concave function of the time average attribute vector, subject to additional time average constraints. The algorithm is based on Lyapunov optimization concepts and involves minimizing a ``drift-plus-penalty'' ratio over each frame. The algorithm can learn efficient behavior without a-priori statistical knowledge by sampling from the past. Our framework is applicable to a large class of problems, including Markov decision problems.

Track 4 – D. Adaptive Systems and Processing

Session: MPb4 – Advances in Adaptive Algorithms

Co-Chairs: ***Sergios Theodoridis,*** *University of Athens* **and *Isao Yamada,*** *Tokyo Institute of Technology*

MP4b-1 3:30 PM

Adaptive Estimation of Sparse Signals using the Method of Multipliers

Daniele Angelosante, Georgios B. Giannakis, University of Minnesota

Adaptive algorithms are well-appreciated for reducing memory requirements and complexity in estimating time-invariant signals, as well as for tracking slow signal variations. In this context, it has been recently established that recursive estimators based on the least-absolute shrinkage and selection operator (Lasso) outperform the celebrated recursive least-squares (RLS) algorithm, when the signal of interest is sparse. On the other hand, sparsity-aware estimators rely on the l1-norm of the signal of interest, and cannot be expressed in closed form, which challenges the adaptation process. To cope with sparsity-aware adaptive real-time processing, this paper develops a recursive estimator based on the alternating direction method of multipliers (ADMoM). Numerical tests demonstrate that the novel ADMoM-based algorithm outperforms existing approaches in terms of mean-square error and complexity while it can efficiently estimate sparse signals.

MP4b-2 3:55 PM

Time- and Coefficient-Selective Diffusion Strategies for Distributed Parameter Estimation

Stefan Werner, Aalto University School of Science and Technology; Yih-Fang Huang, University of Notre Dame

In diffusion-based distributed parameter estimation, the most energy-consuming operation is the communication and diffusion of local estimates among neighboring sensor nodes. It is thus most beneficial to reduce the frequency and the number of parameter estimates to diffuse. This paper proposes an innovative algorithm, wherein the nodes update, hence diffuse, only a subset of their local parameter vector components at selective time instants. The proposed approach combines the benefits of set-membership adaptive filtering, which includes data-dependent updates of estimates, and those of partial update adaptive filtering, which updates only a subset of the components in a parameter vector. This proposed approach offers a significant reduction in energy consumption in the sensor nodes, for when there is no update, there is no need to communicate or diffuse local estimates.

MP4b-3 4:20 PM

Tracking Behavior of Mobile Adaptive Networks

Sheng-Yuan Tu, Ali H. Sayed, University of California, Los Angeles

Adaptive networks consist of a collection of nodes with learning abilities that interact with each other locally in order to solve distributed processing and distributed inference problems in real-time. Various algorithms and performance analyses have been put forward for such networks, such as the adapt-then-combine (ATC) and combine-then-adapt (CTA) diffusion algorithms, the probabilistic diffusion algorithm, and diffusion with adaptive weight links. In this paper, we add mobility as another dimension and study the behavior of the network when the nodes move in pursuit of a target. Mobility leads naturally to an adaptive topology with changing neighborhoods. Mobility also imposes physical constraints on the proximity among the nodes and on the velocity and location of the center of mass of the network. We develop adaptation algorithms that exhibit self-organization properties and apply them to the modeling of collective behavior in biological systems, such as fish schooling, bird flocking, and honeybee swarming.The results help provide, for example, an explanation for the agile adjustment of network patterns of fish schools in the presence of prey or predators.

MP4b-4 4:45 PM

Low Complexity Projection-based Adaptive Algorithm for Sparse System Identification and Signal Reconstruction

Konstantinos Slavakis, University of Peloponnese; Sergios Theodoridis, University of Athens; Isao Yamada, Tokyo Institute of Technology

This paper presents a novel projection-based adaptive algorithm for sparse system identification and signal reconstruction. The online observed data are utilized to form non-differentiable costs which measure the loss of the regression task in hand. Sparsity is infused in the design by the consideration of the weighted ℓ_1 cost criterion. Overall, a sequential non-smooth convex optimization task is posed. Following very recent advances in projection-based adaptive algorithms, the basic recursion of the algorithm is built upon a simple geometric representation; first, a weighted sum of projection mappings onto a set of hyperslabs is taken, and then the subgradient projection mapping, with respect to the weighted ℓ_1 cost, is applied. Such a geometric

analogue results into an algorithm of low complexity, i.e., linear with respect to the number of unknowns, improving, thus, the computational load of a previously introduced projection-based algorithm. Numerical results are also given to validate the proposed method against very recently developed sparse LMS and RLS type of algorithms.

Track 8 – H. Speech, Image and Video Processing

Session: MPb7 – Advances in Keyword Spotting

Chair: ***Mark A. Clements,*** *Georgia Institute of Technology*

MP7b-1 3:30 PM

Speech/Audio Indexing and Retrieval: Improving Precision

Marsal Gavaldà, Nexidia Inc; Mark Clements, Georgia Institute of Technology; Robert Morris, Maria Koulikov, Peter Cardillo, Jon Arrowood, Nexidia Inc

When indexing and retrieval of voice and audio is based on detection of events, the operating point can be set to specific values to control the tradeoff between missed detections and false alarms. Many strategies can be employed to increase detection rates and/or reduce false alarm rates. This paper describes a set of techniques, such as segment filtering, automatic thresholding, and pronunciation optimization, along with their efficacies.

MP7b-2 3:55 PM

Phonological Feature Based Analysis for Keyword Recognition

Abhijeet Sangwan, John H.L. Hansen, CRSS: Center for Robust Speech Systems

In this paper, we explore the use of Phonological Features as a means of analyzing language structure, and how it can be used to improve speech recognition. Traditional automatic speech recognition (ASR) strategies focus on HMMs that reflect a phoneme sequence as “beads on a string”. Such a strategy limits the effective speech processing when considering migration of ASR systems to new speakers, accents, or dialects of a language, since modeling the vast array of pronunciation differences is not an easy task. By employing phonological feature based framework, we are able to characterize production differences between speaker groups, and thereby increase performance for keyword/key-phrase recognition for audio streams of interest. This study will present the (i) fundamental framework, (ii) analysis of production traits (which are deemed “errors” in HMM techniques but in fact are valid production variations), and (iii) speech recognition results based on keyword/key-phrase recognition experiments.

MP7b-3 4:20 PM

Word-Subword Based Keyword Spotting with Implications in OOV Detection

Jan Cernocky, Igor Szoke, Mirko Hannemann, Stefan Kombrink, Brno University of Technology

Main-stream systems for keyword spotting and spoken term detection are based on the series of Large Vocabulary Continuous Speech Recognizer with subsequent search in its output. These systems are limited by the vocabulary of the recognizer and are not able to detect Out of Vocabulary (OOV) words. This talk will present our work in designing hybrid word-subword keyword spotting systems, that maintain the accuracy of LVCSR, while allowing for detecting OOVs as sequences of sub-word units. We will also show the links of this work to the detection, description and clustering of OOVs, as investigated in the framework of the EC-sponsored project DIRAC.

MP7b-4 4:45 PM

Enhanced Open Vocabulary Spoken Term Detection

Bhuvana Ramabhadran, IBM T. J. Watson Research Center

Information retrieval and spoken-term detection from audio such as broadcast news, telephone conversations, conference calls, and meetings are of great interest to the academic, government, and business communities. Motivated by the requirement for high-quality indexes, this paper explores the effect of using both word and sub-word information to find in-vocabulary and OOV query terms and its effectiveness when combined with a text-based information extraction system. It also explores the trade-off between search accuracy and the speed of audio transcription, addresses cases where search term(s) of interest (queries) are acoustic examples, provided either by identifying a region of interest in a speech stream or by speaking the query term. We provide results comparing different representations and generation mechanisms for both queries and indexes built with word and combined word and subword units, evaluate the impact of automated speech recognition error detection and OOV in improving precision. The infrastructure is based on a novel, vocabulary independent, hybrid LVCSR approach to audio indexing and search.

Track 1 – A. Communications Systems

Session: TAa1 – Network Error Correction and Physical Layer Security

Chair: ***Joerg Kliewer,*** *New Mexico State University*

TA1a-1 8:15 AM

Interactive Protocols for Secure Network Coding

Mahdi Jafari, Christina Fragouli, Ecole Polytechnique Fédérale de Lausanne; Suhas Diggavi, University of California, Los Angeles

Network coding allows to improve several aspects of information transfer through networks, that range from increased throughput to higher reliability. In this work we will focus on secure information transfer using network coding techniques. We will propose and analyze protocols that leverage interactive communication to guarantee a target secrecy rate with low computational complexity.

TA1a-2 8:40 AM

Network Coding on Planar Networks under Node-Based Byzantine Attack

Oliver Kosut, Lang Tong, Cornell University; David Tse, University of California, Berkeley

Network coding is studied under Byzantine attack. In particular, an unknown subset of nodes in the network is controlled by an adversary, yet reliable performance is still desired. It has been shown that an attack on nodes is considerably more difficult to deal with than an attack on edges, the latter case having been solved by Cai and Yeung. It is shown that a cut-set upper bound can be achieved for a class of planar networks. This requires the use of a coding strategy known as the Polytope Code, which is a class of nonlinear codes operating over polytopes in real vector fields. This polytope structure induces properties on marginal distributions of code vectors so that validities of codewords can be checked by internal nodes of the network. The planarity condition on the network ensures that there are numerous opportunities to perform checks at internal nodes, allowing the network to fully take advantage of the Polytope Code.

TA1a-3 9:05 AM

Capacity Reservation Algorithms for Mitigating Byzantine Failures in Communication Networks

Khushboo Kanjani, Mohammad Asad Chaudhry, Alex Sprintson, Texas A&M University

We consider the problem of establishing reliable unicast and multicast connections over communication networks with Byzantine link failures. It was recently shown that the novel technique of network coding facilitates instantaneous recovery from link failures and minimizes the amount of additional capacity that needs to be reserved on network links. In this paper, we provide an algorithm for reserving resilient capacity required to tolerate any single link failure in the network. Since the problem is NP-hard, we give an approximation algorithm with provable performance guarantees. The performance of our algorithms is also verified through an extensive simulation study.

TA1a-4 9:30 AM

Network RS codes: Efficient Byzantine Adversary Localization

Hongyi Yao, California Institute of Technology; Sidharth Jaggi, Minghua Chen, Chinese University of Hong Kong

This work introduces Network Reed-Solomon codes (N-RSC) for the multicast problem, which have the following properties: 1. N-RSC are distributed linear network codes that are robust against dynamic network updates. 2. N-RSC achieve the network multicast capacity with high probability. 3. In (adversarially or noisily) faulty networks N-RSC enable error localization in a computationally efficient manner without knowledge of the network topology. In particular, if each internal node has at least d outgoing edges, the identity of up to d/2 faulty network edges can be determined, which is the optimal result one can achieve for networks using linear network codes, even with the knowledge of the network topology.

Track 1 – A. Communications Systems

Session: TAa2 – Signal Processing for Communications Receivers

Chair: ***Mats Viberg,*** *Chalmers University of Technology*

TA2a-1 8:15 AM

Decision Feedback Equalization With Sparsity Driven Thresholding

Jovana Ilic, Thomas Strohmer, University of California, Davis; Raymond Guan, Intel Corporation

For single carrier systems with frequency domain equalization, decision feedback equalization (DFE) performs better than linear equalization, and has a lower complexity and computational costs than optimum equalizers. The main challenges in DFE is the feedback symbol selection rule. In this paper, we give a theoretical framework for a simple, sparsity based thresholding algorithm. We feed back multiple symbols in each iteration, so the algorithm converges fast, and has a low computational cost. The algorithm is applicable in several existing wireless communication systems (SC-FDMA, MC-CDMA, MIMO-OFDM). Our numerical results illustrate significant performance improvement in terms of bit error rate compared to the MMSE solution.

TA2a-2 8:40 AM

The Effect of Unreliable LLR Storage on the Performance of MIMO-BICM

Clemens Novak, Vienna University of Technology; Christoph Studer, Andreas Burg, ETH Zurich; Gerald Matz, Vienna University of Technology

We investigate soft-out demodulators delivering quantized LLRs in MIMO systems employing bit-interleaved coded modulation. The quantized LLRs are mapped to binary labels and stored in unreliable memory introducing i.i.d. bit flips. We investigate the impact of this intermediate LLR channel on the performance of the overall systems in terms of mutual information and compare different binary number representations and labeling strategies. Finally, we propose to encode the labels by an error correcting code to improve performance.

TA2a-3 9:05 AM

On Performance Prediction of an Iterative Multi-Antenna Receiver

Jarkko Huusko, Juha Karjalainen, Markku Juntti, University of Oulu

The statistical properties of a frequency-selective single-input multiple-output channel are used to predict the signal-to-interference-and-noise ratio (SINR) at the output of an iterative equalizer. The SINR is then used to calculate the variance of the bit log likelihood ratios (LLRs). Extrinsic information transfer (EXIT) charts establish a relationship between the LLR variance and mutual information. Thus, by comparing the semi-analytically calculated statistical distribution of the LLR variance at the equalizer output with a previously measured EXIT chart of a turbo decoder, we can reliably estimate of the expected frame error ratio. This prediction method is extended for Chase combining hybrid automatic repeat-request protocol.

TA2a-4 9:30 AM

Doppler Estimation and Correction for Shallow Underwater Acoustic Communications

Kenneth A. Perrine, Karl F. Nieman, Terry L. Henderson, Keith H. Lent, Terry J. Brudner, Brian L. Evans, University of Texas at Austin

Reliable mobile underwater acoustic communication systems must compensate for strong, time-varying Doppler effects. Many Doppler correction techniques rely on a single bulk correction to compensate first-order effects. In many cases, residual higher-order effects must be tracked and corrected using other methods. The contributions of this paper are evaluations of (1) signal-to-noise ratio (SNR) performance from three Doppler estimation and correction methods and (2) communication performance of Doppler correction with static vs. adaptive equalizers. The evaluations use our publicly available shallow water experimental dataset, which consists of 360 packet transmission samples (each 0.5s long) from a five-channel receiver array.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TAa3 – Recursive Reconstruction of Sparse Sequences

Chair: ***Namrata Vaswani,*** *Iowa State University*

TA3a-1 8:15 AM

On the Role of the Properties of the Non-zero Entries on Sparse Signal Recovery

Yuzhe Jin, Bhaskar D. Rao, University of California, San Diego

We study the role of the nonzero entries on the performance limits in support recovery of sparse signals. The key to our results is the recently studied connection between sparse signal recovery and multiple user communication. By leveraging the concept of outage capacity in information theory, we explicitly characterize the impact of the probability distribution imposed on the nonzero entries of the sparse signal on support recovery. When Multiple Measurement Vectors (MMV) are available, we show that the identification of the nonzero rows of the signal is closely connected to decoding the messages from multiple users over a Multiple-Input Multiple-Output channel. Necessary and sufficient conditions for support (indices of nonzero rows) recovery are provided, and the results allow us to understand the role of (row) correlation of the non-zero entries as well as the role of the rank of the matrix formed from the non-zero entries.

TA3a-2 8:40 AM

Video Concealment via Matrix Completion at High Missing Rates

Trac Tran, Johns Hopkins University

abstract

TA3a-3 9:05 AM

Exact Reconstruction Conditions and Error Bounds for Regularized Modified Basis Pursuit (Reg-Modified-BP)

Wei Lu, Namrata Vaswani, Iowa State University

In recent work, we studied the problem of sparse reconstruction with partial, and possibly partly erroneous, knowledge of the support. Denote this support knowledge by T. The idea of our proposed solution, modified-CS (or actually modified basis pursuit), was to search for the signal that is sparsest outside of the set T and satisfies the data constraint. If some prior knowledge of the signal values on T is also available, this can be used to further improve reconstruction error by solving a regularized modified-BP problem. In this work, we will quantify the performance of regularized modified-CS.

TA3a-4 9:30 AM

Iterative Weighted l_1 Optimization for Compressed Sensing and Coding

Amin Khajehnejad, Alex Dimakis, Babak Hassibi, California Institute of Technology

abstract

Track 6 – F. Biomedical Signal and Image Processing

Session: TAa4 – Shape and Time in Biomedical Images

Chair: ***David Shattuck,*** *UCLA Geffen School of Medicine*

TA4a-1 8:15 AM

Spatio-Temporal Image Analysis for Longitudinal and Time-Series Image Data

Guido Gerig, University of Utah

Clinical imaging increasingly makes use of longitudinal image studies to examine subject-specific changes or organ motion due to pathology, intervention, therapy, neurodevelopment, or neurodegeneration, resulting in 4D image data. Cross-sectional analysis of longitudinal data does not provide a model of growth or change that considers the inherent correlation of repeated images of individuals. We will present work in progress towards novel spatio-temporal image processing methods to analyze dynamic growth of organ shape and appearance change in individuals and subject populations.

TA4a-2 8:40 AM

Imaging and Shape Analysis of the Moving Human Fetal Brain In-Utero

Colin Studholme, University of California, San Francisco

This paper describes techniques that allow the formation and analysis of high resolution 3D MR images of the developing human fetal brain in utero, by carrying out object based motion correction of the moving fetal head within deforming maternal tissues. We will then describe an approach to form a complete 4D model of brain shape, MRI contrast and tissue probability over time in the early stages of brain surface folding. A key application of this to template driven EM segmentation of fetal brain tissues will be described, together with applications to morphometric studies of brain growth.

TA4a-3 9:05 AM

Extraction of Functional Subnetworks in Autism Using Multimodal MRI

James Duncan, Michael An, Lawrence Staib, Kevin Pelphrey, Yale University

Autism, Asperger's syndrome, and Pervasive Developmental Disorder - Not Otherwise Specified (referred to collectively as autism spectrum disorder or ASD) are devastating neurodevelopmental disorders. The discovery of reliable image-derived biomarkers could potentially identify people with ASD, or infants who will subsequently develop or are already developing subtle signs of ASD. We hypothesize that the quantification of regional signal change and connectivity information from particular functional subnetworks in the brain responding to ASD-related biological motion tasks, informed by anatomical and diffusion information, will provide more sensitive and robust image-derived biomarkers for studying ASD. Thus, we focus our efforts on the development of a unique mathematical approach that uses image-derived information regarding gray matter location and white matter pathways to inform the estimation of three functionally connected subnetworks related to ASD. Our computational strategy is aimed at grouping ASD-task-related activations into these functional subnetworks using a new multi-view Estimation Maximization (EM) formulation.

TA4a-4 9:30 AM

Multivariate Statistical Analysis of Deformation Momenta Relating Anatomical

Sarang Joshi, University of Utah

The purpose of this study is to characterize the neuroanatomical variations observed in neurological disorders such as dementia. We do a global statistical analysis of brain anatomy and identify the relevant shape deformation patterns that explain corresponding variations in clinical neuropsychological measures. The motivation is to model the inherent relation between anatomical shape and clinical measures and evaluate its statistical significance. We use Partial Least Squares for the multivariate statistical analysis of the deformation momenta under the Large Deformation Diffeomorphic framework. The statistical methodology extracts pertinent directions in the momenta space and the clinical response space in terms of latent variables. We report the results of this analysis on 313 subjects from the Mild Cognitive Impairment group in the Alzheimer's Disease Neuroimaging Initiative (ADNI).

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TA5 – Compressive Sensing

Chair: ***Ali Pezeshki,*** *Colorado State University*

TA5-1 8:15 AM

Target Estimation Using Compressive Sensing for Distributed MIMO Radar

Sandeep Gogineni, Arye Nehorai, Washington University in St. Louis

Distributed Multiple Input Multiple Output (MIMO) radar systems enable viewing the targets from different angles, thereby providing spatial diversity gain. In this paper, we propose an approach to accurately estimate the parameters (position, velocity) of multiple targets using such systems from fewer number of samples by employing compressive sensing. We also introduce a new metric to analyze the performance of the radar system. We show the improvement in performance over conventional Single Input Single Output (SISO) radar systems due to the spatial diversity offered by MIMO radar. We also demonstrate that the sampling rates can be significantly reduced by using compressive sensing at the receivers.

TA5-2 8:40 AM

Sparse Signal Recovery with Dynamic Update of Overcomplete Dictionary

M. Salman Asif, Justin Romberg, Georgia Institute of Technology

Sparse signal priors help in a variety of modern signal processing tasks. In a typical sparse recovery problem, a sparse signal needs to be recovered from an underdetermined system of equations. For example, sparse representation of signal in an overcomplete dictionary or reconstruction of a sparse signal from a small number of linear measurements. In recent years, several results have been presented which guarantee reliable reconstruction under certain conditions. In this paper we investigate the problem of recovering sparse signals when elements are added to (or removed from) the overcomplete dictionary. We propose a dynamic update algorithm for basis pursuit denoising (BPDN), when new columns are added to (or removed from) the dictionary. We use this update procedure to iteratively update the working set of basis elements, chosen from a large library of basis elements, to compute the sparse solution. We also discuss the extension of the same ideas for the analysis-based BPDN.

TA5-3 9:05 AM

Robust Layered Sensing: From Sparse Signals to Sparse Residuals

Vassilis Kekatos, Georgios B. Giannakis, University of Minnesota

One of the key challenges in sensing networks is the extraction of information by fusing data from a multitude of possibly unreliable sensors. Robust sensing, viewed here as the simultaneous recovery of the wanted information-bearing signal vector together with the subset of (un)reliable sensors, is a problem whose optimum solution incurs combinatorial complexity. The present paper relaxes this problem to its closest convex approximation that turns out to yield a vector-generalization of Huber's scalar criterion for robust linear regression. The novel generalization is shown equivalent to a second-order cone program, and exploits the block-sparsity inherent to a suitable model of the residuals. A computationally efficient solver is developed using a block-coordinate descent algorithm, and is tested with simulations.

TA5-4 9:30 AM

Tracking and Smoothing of Time-Varying Sparse Signals via Approximate Belief Propagation

Justin Ziniel, Lee C. Potter, Philip Schniter, Ohio State University

A vast body of literature addressing the problem of sparse signal recovery has emerged in recent years under the banner of Compressive Sensing. The predominant focus of work to date has been on recovering a single unknown vector that is sparse in a suitable basis using a small collection of measurements. Comparatively little attention has been given to the problem of recovering a sequence of sparse vectors that have evolved over time in a structured manner. In this work, we propose a novel algorithm that addresses this task using Markov modeling and approximate belief propagation.

BREAK 9:55 AM

TA5-5 10:15 AM

Performance Analysis of Stochastic Signal Detection with Compressive Measurements

Thakshila Wimalajeewa, Hao Chen, Pramod K. Varshney, Syracuse University

Compressive sampling/sensing (CS) enables the recovery of sparse or compressible signals from a relatively small number of randomized measurements compared to \emph{Nyquist-rate} samples. Although most of the CS literature has focused on sparse signal recovery, exact recovery is not actually necessary in many signal processing applications. Solving inference problems with compressive measurements has been addressed by recent CS literature. This paper takes some further steps to investigate the potential of CS in signal detection problems. We provide theoretical performance limits verified by simulations for detection performance in random signal detection with compressive measurements.

TA5-6 10:40 AM

Compressed Sensing of Different Size Block-Sparse Signals: Efficient Recovery

Ali Ziaei, Ali Pezeshki, Saeid Bahmanpour, Mahmood Reza Azimi-Sadjadi, Colorado State University

This paper considers compressed sensing of different size block-sparse signals, i.e. signals with nonzero elements occurring in blocks with different lengths. A new sufficient condition for mixed l2\l1-optimization algorithm is derived to successfully recover k-sparse signals. We show that if the signal possesses k-block sparse structure, then via mixed l2\l1- optimization algorithm, a better reconstruction results can be achieved in comparison with the conventional l1-optimization algorithm and fixed-size mixed l2\l1-optimization algorithm. The significance of the results presented in this paper lies in the fact that making explicit use of different block-sparsity can yield better reconstruction properties than treating the signal as being sparse in the conventional sense, thereby ignoring the structure in the signal.

TA5-7 11:05 AM

Analog Sparse Approximation for Compressed Sensing Recovery

Christopher Rozell, Georgia Institute of Technology; Pierre Garrigues, IQ Engines, Inc.

Non-smooth convex optimization programs such as L1 minimization produce state-of-the-art results in many signal and image processing applications. Despite the progress in algorithms to solve these programs, they are still too computationally expensive for many real-time applications. Using recent results describing dynamical systems that efficiently solve these types of programs, we demonstrate through simulation that custom analog ICs implementations of this system could potentially perform compressed sensing recovery for real time applications approaching 500 KHz. Furthermore, we show that this architecture can implement several other optimization programs of recent interest, including reweighted L1 and group L1 minimization.

TA5-8 11:30 AM

High Resolution Radar via Compressive Illumination

Emre Ertin, Ohio State University

Frequency Modulated Continuous Wave (FMCW) radars achieve high range resolution using frequency modulated. The Nyquist sampling rate for the ADC is determined by the radar spot size and chirp rate which is typically less than the total bandwidth swept by the linear FM signal. Sub-Nyquist sampling in time has been avoided in practice since it creates potential ambiguities in range. Recent results collectively known as compressive sensing has provided provable performance guarantees and signal recovery algorithms for random sub-sampling of sparse or compressible signals. In this paper we propose a novel compressive sensing strategy for radar that relies on on using compressive illumination with waveform designs across frequency, that shifts the burden of the sampling operator from the receiver to the transmitter. Aliasing after adaptive illumination reduces the sampling bandwidths at the expense of increased complexity in the transmitter structures. We present theoretical analysis of the compressive illumination strategy through characterization of the coherency of the resulting sampling dictionary and relation between bandwidth, sampling rate and scene sparsity. We also provide results on experimental demonstration of the compressive illumination strategy through sampling of staggered multifrequency linear FM signals through a single low rate A/D.

Track 7 – G. Architecture and Implementation

Session: TAa6 – Reconfigurable Architectures, Algorithms and Applications

Chair: ***F. de Dinechin,*** *Ecole Normale Supérieure de Lyon*

TA6a-1 8:15 AM

A Generic and Versatile Architecture for Inference of Evolutionary Trees under Maximum Likelihood

Nikolaos Alachiotis, Alexandros Stamatakis, Technische Universität München

Likelihood-based reconstruction of evolutionary trees from molecular sequence data exhibits extreme resource requirements because of the high computational cost of the phylogenetic likelihood function. We propose a dedicated computer architecture for significantly accelerating the execution of likelihood-based phylogeny programs. This design is sufficiently generic to support any possible input data type, e.g., DNA, RNA secondary-structure, or protein data. Furthermore, the architecture is able to calculate log-likelihood scores, perform scaling operations to ensure numerical stability on large datasets, optimize the branch lengths of the tree topologies, and calculate instantaneous transition probability matrices. We verify our architecture by using FPGA technology.

TA6a-2 8:40 AM

Is there a Tradeoff Between Programmability and Performance

Walid Najjar, University of California, Riverside; Jason Villarreal, Jacquard Computing Inc.

FPGAs are commonly used as execution platforms for signal and image processing applications because they provide a good tradeoff between the programmability of CPUs and DSPs and the performance of ASICs.However, their programmability remains a major barrier to their wider acceptance by application code developers. These platforms are typically programmed in a low level hardware description language, a skill not common among application developers and a process that is often tedious and error-prone. The Riverside Optimizing Compiler for Configurable Circuits (ROCCC) is a C-to-VHDL compilation toolset designed to raise the abstraction of FPGA programming. Two important features of ROCCC are (1) support for modular code design and (2) support for extensive program optimizations and transformations. The first feature allows the programmer to build an application code using the composition of independent and separately designed and tested modules as building blocks. These modules are stored in a database integrated with the compiler and managed by the Eclipse-base GUI. A module can be imported into another module in one of three forms: as C code, as VHDL code or as a netlist. The second feature allows the programmer to apply a wide variety of program transformations designed to exploit the available parallelism on the chip, reduce the area occupied on the chip, reduce the number of off-chip memory accesses and improve the clock frequency of the circuit.

Many applications in signal, image and video processing consist of a sequence of well defined and often standard operators applied to a data set or a stream of data. While it is possible to apply these compiler transformations to each operator separately, studies have shown that applying them to the whole application yields dramatically better results in terms of speed, area and throughput. By allowing the user to import modules under three forms, ROCCC not only improve the productivity by supporting code reuse, it also reduces design variability by allowing the use of tested cores in the forms of netlists. However, importing a module as a netlist or a VHDL code prevents the compiler from applying the above mentioned program transformations to the whole application. In this paper we evaluate the tradeoffs, costs and opportunities provided by combining modular building block application design when combined with compile-time program transformations. We use a large set of signal, image and video applications and kernels to evaluate the tradeoff.

TA6a-3 9:05 AM

FPGA-Optimised Random Number Generators

David Thomas, Wayne Luk, Imperial College

Using FPGAs to accelerate Monte-Carlo (MC) simulations has huge potential: the intrinsic parallelism of MC makes it relatively easy both to develop pipelined simulation cores, and to scale the number of cores across a device. However, a key component of any simulation is the Random Number Generator (RNG) providing the underlying source of randomness, which must appear as statistically random as possible. There has been extensive research into efficient software RNGs, but these are often less efficient when translated to hardware, while traditional hardware RNGs such as LFSRs cannot provide the quality needed in long-running simulations. Recent research into FPGA-optimised RNGs has produced a number of methods which are specifically targeted at the features of FPGAs, such as LUTs and FIFOs. This paper will examine developments in three types of RNG: uniform bit generators, exponential variate generators, and multivariate Gaussian generators. In all three cases the bit-wise nature of FPGAs enables RNG architectures radically different from those found in software, while maintaining significant quality. This means that while RNGs are seen as an expensive multi-instruction operation in software, in an FPGA the RNG is usually cheaper than a single floating-point operation.

TA6a-4 9:30 AM

A 128-tap Complex FIR Filter Processing 20 Gigasamples/s in a Single FPGA

Florent de Dinechin, Honoré Takeugming, École Normale Supérieure de Lyon; Jean-Marc Tanguy, Bell Laboratories, Alcatel-Lucent

To enable 40Gb/s data transmission over older optical fibres using QPSK modulation, the first step of the receiver signal-processing pipeline is a long (128-tap) Finite Impulse Response filter that compensates the chromatic dispersion due to the medium. We present an implementation of this FIR filter that is able to process 20 giga-samples per second, where each sample is a complex number with 5+5 bits resolution. Our implementation, targetted to the largest Stratix-IV GX device, processes 128 complex samples per cycles at a frequency of 156MHz thanks to an FFT-based architecture. The FFT and iFFT pipelines use ad-hoc memory-based constant multipliers well suited to the FPGA features, while the multiplications in the Fourier domain use the FPGA embedded DSP blocks. This FPGA is thus able to perform more than 2 tera-operations per second. The precision of the intermediate signals is chosen to ensure that the error of the output signal with respect to the Matlab reference is never more than one least significant bit.

Track 8 – H. Speech, Image and Video Processing

Session: TA7 – Image and Video Enhancement

Chair: ***Manu Parmar,*** *Qualcomm, Inc.*

TA7-1 8:15 AM

Camera Technology at the Dawn of Digital Renascence Era

Sergio Goma, Mickey Aleksic, Qualcomm Inc.; Todor Georgiev, Adobe Systems

Camera Technology has evolved tremendously in the last 10 years, the proliferation of camera-phones fueling unprecedented advancements on CMOS image sensors. As an emerging field, some of the problems are justified while others are bi-products of the chosen silicon technology and are not fundamental to advancement of image technology. This paper will review block by block the image processing components found in a cell-phone camera today, discussing for each its justification from technology choice vs image processing function, with emphasis on the signal degradation potential. Further, we present computational photography challenges that amplify the requirements for data with high SNR. A discussion of image quality evaluations concludes this presentation.

TA7-2 8:40 AM

Rethinking the Sampling Topologies for Image Quality Estimation in Computational Imaging System Design

Kathrin Berkner, Ricoh Innovations, Inc

Optical design includes an optimization process that evaluates image degradations caused by the lens design in each iteration. In joint digital-optical imaging systems design the optimization cost function needs to include measured image quality after optical and digital processing. A commonly used merit function is the Mean-Squared-Error (MSE) of the digital image after restoration. Low resolution rectangular sampling grids increase the speed of the optimization process, but cause approximation errors of the MSE merit function. Through adaptation of the MSE calculations to flexible non-rectangular sampling topologies, we achieve increased MSE accuracy and significant speed-up of the processing time

TA7-3 9:05 AM

Novel YUV 8bpp Subsampling Pattern

Sergio Goma, Mickey Aleksic, Qualcomm Inc.

We propose a novel 8bpp subsampling YUV pattern based on a checkerboard subsampling of the luminance component that explicitly preserves the edge. The proposed pattern uses 1bit to encode edge direction in the missing luminance pixel and this bit is stored in the chroma sample as the chroma sample is DPCM encoded 8 to 7bits per sample. The complexity analysis of both encoder and decoder is concluded with a proposed hardware implementation. The image quality performance of the proposed pattern is estimated using MTF measurements, quantifying the loss in high-frequencies and a comparison is presented across the YUV subsampling methods.

TA7-4 9:30 AM

Robust Image Registration for Multi-frame Mobile Applications

Marius Tico, Kari Pulli, Nokia Research Center

We introduce a new approach to global image registration that allows a robust and efficient image alignment, required in various imaging applications (e.g. digital image stabilization, high dynamic range imaging, panoramic image stitching). The proposed method follows a coarse to fine strategy, adopting different registration techniques at coarse and fine resolution levels. Coarse levels are registered using a featureless approach, whereas fine levels are registered based on salient features detected in the two images.

BREAK 9:55 AM

TA7-5 10:15 AM

Quality-controlled Motion-compensated Interpolation

Mina Makar, Derek Pang, Yao-Chung Lin, Bernd Girod, Stanford University

Low-complexity video encoding is essential for mobile and power-sensitive applications. One way to reduce encoding complexity is to drop frames at the encoder and perform motion-compensated frame interpolation at the decoder. This often results in interpolation artifacts. We propose a method to estimate the quality of the interpolated frames at the decoder by transmitting a small amount of error control information in lieu of an omitted frame. Typically, this information is obtained from a projection of the frame on a suitable low-dimensional basis. The projection coefficients can be compressed by conventional techniques or, more efficiently, by Slepian-Wolf coding. Based on the quality estimate, the decoder can recognize and suppress occasional frames for which motion-compensated interpolation does not yield satisfactory picture quality. Experimental results demonstrate that our approach eliminates most interpolation artifacts and achieves much better visual quality at a negligible increase in bit-rate.

TA7-6 10:40 AM

A Constrained Optimization Perspective on Joint Spatial Resolution and Dynamic Range Enhancement

Vishal Monga, Umamahesh Srinivas, Pennsylvania State University

Recent research has seen a surge of approaches for enhancing spatial as well as amplitude (or dynamic range) resolution in images from multiple captures. We develop a novel constrained optimization framework which addresses the problem of joint estimation of imaging model parameters (registration and blur) and the unknown hi-res image. In this framework, we employ a transformation of variables to establish separable convexity of the cost function (as well as constraints) under any lp norm, p >= 1, in the individual variables of geometric and photometric registration parameters, optical blur and the unknown hi-res image. The convergence guarantees afforded by our algorithm alleviates unreasonable demands on initialization, and produces reconstructed image results approaching practical upper bounds. Several existing formulations reduce to special cases of our framework making the algorithm broadly applicable.

TA7-7 11:05 AM

Bleed-Through Removal Using Multispectral Image Data

Trace Griffiths, Gene A. Ware, Todd Moon, Jacob Gunther, Utah State University

Removing bleed-through texts found in digitally scanned or imaged documents is a current need. Many algorithms have been presented to correct bleed-through using grayscale and/or color images. Recent application of multispectral imaging techniques to documents increases the signal information available for bleed-through correction. We utilize the multispectral information from both sides of the document and principal component analysis to correct bleed-through texts in a fast, unsupervised algorithm.

Track 3 – C. Networks

Session: TAa8 – Cognitive Networking 8:15 AM – 9:55 AM

Chair: ***Georgios Giannakis,*** *University of Minnesota*

TA8a2-1

Cooperative Wideband Spectrum Sensing Using Radio Frequency Sensor Networks

Volkan Sonmezer, Turkish Air Force; Murali Tummala, John McEachen, Naval Postgraduate School

This paper implements spectrum sensing using a radio frequency sensor network and analyzes the performance of this implementation through simulation. A sensor network based cooperative wideband spectrum sensing scheme is proposed for the implementation of the task. In the proposed scheme, wavelet-based multi-resolution spectrum sensing, which was originally proposed for cognitive radio applications, is adapted to radio frequency sensor networks. For cooperation of the nodes in the proposed scheme, a new three-bit hard combination technique is developed. A simulation model is created in MATLAB programming language to implement the proposed scheme and to analyze its simulation performance. The results of the simulation show that the proposed sensor network based cooperative wideband spectrum sensing scheme is appropriate for radio frequency sensor networks and the proposed three-bit hard combination scheme is superior to the traditional hard combination schemes in terms of false alarm reduction.

TA8a2-2

Spectrum Leasing via Cooperative Opportunistic Routing

Davide Chiarotto, University of Padova; Osvaldo Simeone, New Jersey Institute of Technology; Michele Zorzi, University of Padova

A spectrum leasing mechanism is proposed for the coexistence between a primary and a secondary network that is based on cooperation and opportunistic routing. The primary network consists of a source and a destination communicating via a number of primary relay nodes. In each transmission block, the next hop is selected in an on-line fashion based on the realized decoding outcomes in the previous transmissions according to the idea of opportunistic routing. The secondary nodes may serve as potential next hops for the primary network, but only in exchange for leasing of spectral resources so as to satisfy secondary quality-of-service constraints. Four policies based on spectrum leasing via opportunistic routing are proposed that provide different trade-offs between gains in throughput and overall energy expenditure for the primary network. Analysis is carried out for networks with a linear geometry and quasi-static Rayleigh fading statistics by using Markov chain tools.

TA8a2-3

Effect of Jamming on Distributed Spectrum Sensing in a Cognitive Radio Network

V Sriram Siddhardh (Sid) Nadendla, Hao Chen, Pramod K Varshney, Syracuse University

We design the optimal jamming attack strategy for a cognitive radio network in the presence of path-loss decaying signal models. We consider a cognitive radio network with K participating cognitive radios and one fusion center in the presence of one primary user and one jammer in the operating region. We assume that the network is not aware of the presence of the jammer and hence employs the optimal decision rules designed for a benign environment. Jammer, on the other hand, tries to take advantage of this ignorance and carries the best possible attack so that it can maximally deteriorate the global performance (error-probability) of the network under a total power-constraint. We consider a two-fold attack - one on the sensor reception and other on the fusion center reception. We present numerical simulations depicting near-field and far-field effects over different path-loss exponents to find the optimal jamming attack for a relatively simple example where the network has only one sensor ($K = 1$). This example serves as an illustration of the basic concepts and will be followed by a more in-depth study.

TA8a2-4
Performance Analysis of Weighted Centroid Algorithm for Primary User Localization in Cognitive Radio Networks
Jun Wang, Paulo Urriza, Yuxing Han, Danijela Čabrić, University of California, Los Angeles

Location information in a cognitive radio (CR) network enables several key improvements in spatio-temporal spectrum sensing, location-aware routing and spectrum policy enforcement. The weighted centroid localization (WCL) algorithm is a robust and low-complexity technique that is suited for the non-interactive localization scenario in CR networks. We present the first theoretical analysis of the error distribution of WCL in both uncorrelated and correlated shadowing. Through this analysis, the closed-form pdf of the localization error is obtained and verified to match simulations. These results are used to provide practical guidelines regarding system design of WCL for CR networks.

TA8a2-5
Optimizing User Densities for Spectrum Allocation with Applications in Femtocell Networks
Brett Kaufman, Rice University; Jorma Lilleberg, Nokia; Behnaam Aazhang, Rice University

In this work we analyze a spectrum allocation problem between two classes of users. We study the problem from the perspective of a cellular network with an underlaid femtocell network. Femtocells are deployed by the end-user, and thus the topology of femtocell networks is highly dynamic and stochastic in nature. One way of managing the interference is to control the population size of the network. By fixing the spectrum resources available to the femtocell network, an optimal density of femtocells can be found that maximizes the performance of the network in terms of the achievable sum rate.

Track 2 – B. MIMO Communications and Signal Processing

Session: TAa8 – Cooperative and Cognitive Transmission in Multi-Antenna Networks I 8:15 AM – 9:55 AM

Chair: ***Kaibin Huang,*** *Yonsei University*

TA8a1-1
Randomized Two-Way Relay Cooperation
Saeed Bagheri, University of California, Davis; Francesco Verde, University Federico II; Donatella Darsena, University of Napoli Parthenope; Anna Scaglione, University of California, Davis

In this work we propose a new decentralized coding method to harness the two way relay multiple access efficiency as well as diversity benefits from a group of uncoordinated relays, via a new scheme that we call ”randomized two way relay cooperation”. The idea is to combine randomized cooperative coding with the two way relay scheme to attain the formation of the two-way relay channel in an ad-hoc manner, with virtually no overhead. The key idea is that through randomized cooperation the source and destination pair can be unaware of the actual set of relays that are contributing to their two way communication and still rip the benefits of the presence of a relay channel that appears has having a fixed number of antennae. We analyze the performance of decode and amplify and forward techniques combined with the randomization and calculate the diversity achievable through the system.

TA8a1-2
Distributed Beamforming for Two-way Relay Networks with Reciprocal Channels
Meng Zeng, Texas A&M University; Rui Zhang, National University of Singapore; Shuguang Cui, Texas A&M University

This paper studies the achievable rate region for a two-way relay network with distributed beamforming under the assumption that the source--relay and relay-source channels are reciprocal. Specifically, we investigate the achievable rate regions for two cases, where the relay cluster is subject to a sum-power constraint or individual-power constraints. We show that the optimal beamforming vectors obtained from solving the weighted sum inverse-SNR minimization (WSISMin) problems are sufficient to characterize the corresponding achievable rate region. Furthermore, we derive the closed forms for those optimal beamforming vectors and propose the partially distributed algorithms to implement the optimal beamforming, where each relay node only needs the local channel information and one global parameter.

TA8a1-3
Balanced Precoding for Decode-and-Forward Based MIMO Relay Communications
Jongyeol Ryu, Wan Choi, Korea Advanced Institute of Science and Technology

This paper proposes a new joint precoding technique in decode-and-forward (DF) based multiple-input multiple-output(MIMO) relay communications. Contrary to previous works, the proposed precoding matrix at a source node is designed by balancing source-to-relay (SR) and source-to-destination (SD) links. For the given precoding matrix at a source node, a relay node optimizes its precoding matrix to maximize the achievable rate. The balanced precoding eliminates a capacity bottleneck of relay communications and significantly improves capacity. It is shown that the proposed precoding scheme is shown to substantially outperforms conventional precoding schemes neglecting the SD link.

TA8a1-4
Superposition Coding Based Cooperative Communication with Relay Selection
Hobin Kim, Pamela C. Cosman, Laurence B. Milstein, University of California, San Diego

We study the layered transmission of a Gaussian source over multiple relays using superposition coding and cooperative MIMO. At first, we analyze the outage probability and performance in terms of average throughput and distortion for decode-and-forward protocols with single layer coding and with superposed two-layer coding. For the superposition coding approach, we consider different power allocations to the base and enhancement layers. Then, we propose a simple protocol which assigns a predetermined number of relays to individual layers instead of repeating the superposition coded source packet at the relay. Finally, we present the numerical results based on the analysis to compare the performance.

TA8a1-5
Optimal Power Allocation in Linearly Coded OFDMA Relay Networks
Honghai Yu, Sumei Sun, Institute for Infocomm Research

In this paper, a linearly coded (LC) relaying scheme is proposed for multiuser relay network in which a number of users communicate simultaneously with a common destination with the help of one relay node, by using orthogonal frequency division multiple access (OFDMA). The proposed linear coding at the relay node enables exploiting the frequency diversity in the relay-to-destination (R-to-D) link. Moreover, closed-form pairwise error probability (PEP) and asymptotically tight bit error rate (BER) for linearly coded orthogonal frequency division multiplexing (LC-OFDM) system are derived so that an accurate end-to-end (source to destination) BER expression can be obtained. Based on this accurate end-to-end BER expression, optimal power allocation strategy between source nodes and relay node is easily developed by convex optimization when the total transmit power is a constraint. Simulation results show that our optimal power allocation consumes only half the power of equal power allocation in high signal-to-noise ratio (SNR) region.

TA8a1-6
Distributed Gain Allocation in Non-Regenerative Multiuser Multihop MIMO Networks
Raphael Rolny, Jörg Wagner, Armin Wittneben, Swiss Federal Institute of Technology Zurich

Consider a set of non-cooperating source-destination pairs that communicate over the same physical channel. It is known that an intermediate stage of relay nodes can improve performance in terms of achievable rate. In this paper, we study distributed gain allocation schemes to maximize the achievable sum-rate in non-regenerative multiuser multihop networks with more than two hops and an arbitrary number of antennas at each node. The relays are constrained to forward linear transformations of their receive signals and calculate their transformation matrices based on locally available channel state information and limited feedback from the destination. We devise gradient based optimization schemes and assess the achievable sum-rates of these schemes by computer simulations.

TA8a1-7
Optimal Spectrum Sharing in MIMO Cognitive Radio Networks via Semidefinite Programming
Ying Jun Zhang, Anthony Man-Cho So, Chinese University of Hong Kong

In cognitive radio (CR) networks with multiple-input multiple-output (MIMO) links, secondary users (SUs) can exploit “spectrum holes” in the space domain to access the spectrum allocated to a primary system. However, they need to suppress the interference caused to primary users (PUs), as the secondary system should be transparent to the primary system. In this paper, we study the optimal secondary-link beamforming that balances between the SU’s throughput and the interference it causes to PUs. In particular, we aim to maximize the throughput of the SU, while keeping the interference temperature at the primary receivers below a certain threshold. Unlike traditional MIMO systems, SUs may not have the luxury of knowing the channel state information (CSI) on the links to PUs. This presents a key challenge for a secondary transmitter to steer interference

away from primary receivers. In this paper, we consider three scenarios, namely when the SU transmitter has complete, partial, or no knowledge about the channels to the PU receivers. In particular, when complete CSI is not available, the interference-temperature constraints are to be satisfied with high probability, thus giving rise to chance constraints that are typically hard to deal with. Our contribution is fourfold. First, by analyzing the distributional characteristics of MIMO channels, we propose a unified homogeneous quadratically constrained quadratic program (QCQP) formulation that can be applied to all three scenarios with different levels of CSI knowledge and with either deterministic or probabilistic interference-temperature constraints. The homogeneous QCQP formulation, though non-convex, is amenable to semidefinite programming (SDP) relaxation methods. Secondly, we show that the SDP relaxation admits no gap when the number of primary links is no larger than 2. A polynomial-time algorithm is presented to compute the optimal solution to the QCQP problem efficiently. Thirdly, we propose a randomized polynomial-time algorithm for constructing a near-optimal solution to the QCQP problem when there are more than 2 primary links. Finally, we show that when the secondary transmitter has no CSI on the links to primary receivers, the QCQP problem reduces to a matrix eigenvalue-eigenvector decomposition problem. In this case, the optimal beamforming solution can be obtained very efficiently without the need to solve the QCQP.

TA8a1-8
Two-way Communications for Cooperative Multiple Source Pairs Through a Multi-antenna Relay

Chin Choy Chai, Chau Yuen, Institute for Infocomm Research

We study joint scheduling and beamforming scheme for multiple source pairs that are exchanging information through the relay, where each source has single antenna and the relay has multiple antennas. Unlike previous works, we jointly consider the ordering and scheduling of different source pairs by taking into account their channel gains information and quality of service (QoS) requirements, as well as the joint design of the transmit and receive beamforming at the multi-antenna relay . The objective is to maximize the system throughput by minimizing the inter-pair interference subject to QoS constraints in terms of the SINR requirements. We propose joint beamforming and scheduling schemes, and compare them with the exhaust searching algorithm. New insights on how to design practical scheduling-based beamformer for multi-antenna relay with multiple information-exchanging source pairs are presented.

TA8a1-9
Maximum Achievable Diversity of Coded MIMO-OFDM Amplify-and-Forward Relaying Systems

Changick Song, Inkyu Lee, Korea University

In this paper, we consider multiple-input multiple-output (MIMO) orthogonal frequency division multiplexing (OFDM) amplify-and-forward relay channels. We analyze the maximum achievable diversity of the coded beamforming scheme in MIMO-OFDM relaying systems, and provide proper code designs and the subcarrier pairing criteria that yield the maximum diversity. The pairwise error probability is analyzed based on the assumption of correlated fadings among subcarriers. For systems with N_t source, N_r relay and N_d destination antennas with proper code design, we show that the maximum achievable diversity order is obtained as $\min(L_hN_tN_r,L_gN_rN_d)$, where L_h and L_g denote the number of channel taps of the first hop and the second hop channel, respectively.

TA8a1-10
Max-Min Weighted SIR in Coordinated Multicell MIMO Downlink System

Desmond W.H. Cai, Tony Q.S. Quek, Institute for Infocomm Research, A*STAR

In wireless MIMO downlink systems, it is important to adapt the transmit and receive power and beamformers to improve the performance of the network. Appropriate choice of the transmit and receive power and beamformers can significantly reduce both the intra-cell and inter-cell interference in the system. This paper studies the optimization of a multicell MIMO downlink system in which each base station serves multiple users, and each user is served by only one base station. First, we propose a power control algorithm which maximizes the minimum weighted SIR for all users with little backhaul cooperation between the base stations. Next, we implement the power control algorithm in an alternate optimization procedure to jointly optimize the power and beamformers. We demonstrate that the proposed joint beamforming and power control algorithm provides significant gains over the non-cooperative system.

TA8a1-11

On the Optimization of Two-way AF MIMO Relay Channel with Beamforming

Namjeong Lee, Korea Advanced Institute of Science and Technology; Chan-Byoung Chae, Bell Laboratories, Alcatel-Lucent; Osvaldo Simeone, New Jersey Institute of Technology; Joonhyuk Kang, Korea Advanced Institute of Science and Technology

In this paper, the joint design of transmit beamformers, receive combiners, and a linear relaying matrix is studied for a two-way amplify-and-forward (AF) relay system equipped with multiple-antennas at sources and relay. A single data stream is transmitted by each source. Due to the non-convexity of the optimization problem, finding the solution that maximizes the sum-rate appears intractable. Hence, solution to the original problem is approximated via the iterative solution of three optimization problems: one for the transmit beamformer, one for the receive combiner, and one for the linear relaying matrix. Since the latter is non-convex, a suboptimal iterative procedure is proposed. Joint optimization is assumed to be performed at the relay, which designs the transceiver thanks to perfect channel state information and informs the sources of the transmit beamformers/receive combiners. To reduce the necessary control information towards the sources, limited feedback is also considered using Grassmannian quantization codebooks. For comparison, distributed implementation strategies that do not require relay-to-sources feedback of the system parameters but require own channel information at the sources are considered. Finally, an upper bound to the achievable sum-rate is provided. The proposed techniques show achievable sum-rate performance very close to the upper bound. Moreover, the limited-feedback implementation shows a slight performance degradation even using only 4 feedback bits.

TA8a1-12

Is Conflict Always Bad? From the Interference Management Perspective

Chan-Byoung Chae, Kai Yang, Simon Yiu, Doru Calin, Bell Laboratories, Alcatel-Lucent

In this paper, we first propose a simple two-cell MIMO solution where each base station only has its own data message. The algorithm is based on limited feedback from the mobile stations. We assume that there is no base station cooperation though backbone; therefore cooperation among base stations is not a requirement. This renders the concept of the proposed solution easier for product implementation. To extend the algorithm to a multi-cell scenario, we next propose a novel physical beam switching method based on the proposed two-cell MIMO solution. Conventionally, all base stations align the beam directions to avoid inter-cell interference. In this paper, however, we propose to use a different beam switching method based on conflict graph. Instead of aligning all the beams, we intentionally create a strong interference term. In doing so, background interference except the strongest interference is significantly reduced and the strongest interference term is further removed/minimized by the proposed two-cell MIMO solution. This results in increasing the received signal-to-noise-plus-interference ratio (SINR). Note that, unlike prior work, our solution creates and utilizes the conflict, which means the conflict is useful. Numerical results confirm that the conflict significantly helps the multi-cell system improving the system throughput.

TA8a1-13

Feasible Rate Improvement Using Common Message Decoding for Multicell Networks

Hayssam Dahrouj, Wei Yu, University of Toronto

The performance of a multicell wireless network is often limited by intercell interference. This paper considers the use of common message decoding for interference management for a downlink multicell system with multiple antennas at the base-stations and single antenna at the mobiles. We utilize an SDP relaxation-based strategy proposed in our previous work to determine the optimal beamforming vectors for both common and private messages, and propose a new heuristic algorithm to characterize the improvement in the feasible rate with common-message decoding. Simulation results show that common message decoding can significantly improve the minimum achievable rate for cell-edge users when base-stations are closely spaced from each other.

TA8a1-14

Switched Diversity Strategies for Dual-Hop Relaying Networks

Fakhreddine Gaaloul, Redha Radaydeh, Mohamed-Slim Alouini, Korea Advanced Institute of Science and Technology

This paper investigates the effect of different switched diversity configurations on the implementation complexity and achieved performance of dual-hop amplify-and-forward (AF) relaying networks. A low-complexity model of the relay station is adopted, wherein single-input single-output antenna configuration is employed. Each of the transmitter and the receiver however employs multiple antennas to improve the overall link performance. Single-phase and two-phase based switching strategies are investigated. Thorough comparisons between the implementation complexity, which is defined in terms of the average number of active antennas, average number of channel estimations and switching, and the achieved diversity gain for different switching strategies are presented. Simulation results are also provided to validate the mathematical development and to verify the numerical computations.

TA8a1-15

Stochastic Feedback Control for Multi-Antenna Interference Channel

Rong Ran, Hong Kong University of Science and Technology; Kaibin Huang, Yonsei University; Vincent K. N. Lau, Hong Kong University of Science and Technology; Dongku Kim, Yonsei University

In a multi-antenna interference channels, the knowledge of channel state information (CSI) at the transmitters is very important for interference mitigations. In existing papers, global and perfect knowledge of CSI has been assumed. However, in practice, there is huge signaling overhead in delivering the CSI from the receivers to the transmitters. In this paper, we consider an event-driven CSI feedback design that optimize the average throughput of the multi-antenna interference channels subject to an average feedback cost constraint. Specifically, we consider a decentralized "on/off" feedback controller where each receiver determines whether to transmit CSI feedbacks of the direct link and/or the cross-links respectively based on the local CSI information only. The resultant decentralized stationary control policy is derived using stochastic optimization theory. By exploiting special problem structure, we derived some structural results of the solution and show that the control policy is threshold-based. The proposed solution has low complexity, decentralized and is Pareto optimal. We also characterize the optimal tradeoff between the average throughput and the average feedback rate in the multi-antenna interference channels under the {noise-limited regime} and the { interference limited regime}. Based on the analysis, we discuss various design insights on whether the feedback cost should be spent on the direct link or the cross-links in different operating regimes.

TA8a1-16

Asymptotic Performance of Linear Receivers in Network MIMO

Jakob Hoydis, Mari Kobayashi, Mérouane Debbah, Supélec

We consider the asymptotic performance of a class of linear receive filters in multiple-input multiple-output (MIMO) multiple access channels (MAC). Under the assumption that the number of transmitters K and receive antennas M grow large at the same rate and that the receiver has only an imperfect estimate of the channel matrix, we derive deterministic equivalents for the signal-to-interference-plus-noise-ratio (SINR) at the filter output. Since our model assumes that the channel matrix has a variance profile, i.e., different matrix entries exhibit different variances, the results are useful for the analysis of network MIMO systems where a user faces different path losses to the cooperative base stations. Our simulation results show that the asymptotic performance predictions give accurate approximations even for small M, K.

Track 4 – D. Adaptive Systems and Processing

Session: TAa8 – Adaptive Signal Processing: Theory and Applications 8:15 AM – 9:55 AM

Chair: ***Todd Moon,*** *Utah State University*

TA8a3-1

Spacecraft Adaptive Control Evaluation

Timothy Sands, Naval Postgraduate School (USAF); Jae Jun Kim, Brij Agrawal, Naval Postgraduate School

This paper evaluates spacecraft adaptive attitude control by examining the contributions of individual components of one common adaptive algorithm. Feedforward and feedback controls are briefly introduced for context, then parameter adaptation and reference trajectories are applied individually to feedforward and feedback controls. The effects of noise are also examined. The various control schemes are simulated to heuristically display the impacts of reference trajectories versus desired trajectories, adaptation versus nonadaptive, and also the effects of adaptation and control gains in addition to sensor noise. The simulations are validated by experimental results on a free-floating three-axis spacecraft simulator actuated by nonredundant single-gimbal control moment gyroscopes.

TA8a3-2

A Novel Block Fast Array RLS Algorithm Applied to Linear Flight Strip-Map SAR Imaging

Roger West, Todd Moon, Jacob Gunther, Utah State University

In this paper, a novel block fast array recursive least squares (BFARLS) algorithm is developed that exploits the block Toeplitz structure of a data matrix by using its block rows as block shifted regressors for efficiently solving a regularized system of equations. A new block hyperbolic Householder transformation is used for the hyperbolic rotations. This new BFARLS algorithm is applied to the maximum likelihood (ML) estimation of ground reflectivity from linear flight strip-map synthetic aperture radar (SAR) data, whose solution involves solving a very large system of equations that has the stated structure. Simulation examples are given that illustrate the algorithm and the complexity of the algorithm is analyzed.

TA8a3-3

An Adaptive IIR Filter with Constraints on the Output Power Level

Walter Kozacky, Tokunbo Ogunfunmi, Santa Clara University

Active Noise Control (ANC) systems can be used for suppressing sinusoidal frequency interference in wideband signals. In some applications it may be desirable to limit the maximum power level delivered by the ANC system to prevent overdriving a transducer or causing output amplifier saturation. This paper develops an algorithm for an adaptive IIR filter that places a constraint on the output power level, while still maintaining a narrow bandwidth to reject only the interfering sinusoidal portion.

TA8a3-4

On the Robust and Efficient Computation of the Kalman Gain for Multichannel Adaptive Filtering with Application to Acoustic Echo Cancellation

Karim Helwani, Herbert Buchner, Sascha Spors, Deutsche Telekom Laboratories, Berlin University of Technology

In this paper we discuss efficient approaches for a robust computation of the spatio-temporal Kalman gain in the context of adaptive multichannel systems. We apply the given approaches to the multichannel acoustic echo cancellation problem as representative for the class of system identification problems. Moreover, we provide a complexity and performance analysis of our approaches and compare the results to the conventional spatio-temporal Kalman gain computation methods.

TA8a3-5

An Interval Method for State Estimation in Biological Systems

Maria Angels de Luis Balaguer, Cranos Williams, North Carolina State University

The estimation of state variables and parameters is a major task in systems biology. In the modeling process, uncertainty has to be included in order to get realistic models. Estimation methods that consider uncertainties, and that guarantee the results have been developed to achieve this goal. They have been successful in different areas. However, to our knowledge, there are no methods that have been developed specifically for biological systems, and the general purposes methods arise problems when they are applied to this field. The purpose of this work was to generate an estimation algorithm that was appropriate for biological systems.

TA8a3-6

A 0.18um CMOS Narrow-band LNA Linearization Using Digital Base-band Post-Distortion

Ifiok Umoh, Talal Al-attar, Tokunbo Ogunfunmi, Santa Clara University

In this paper, a novel digital base-band post-distorter is proposed to compensate for nonlinearities in low noise amplifiers. Though high gain, good input matching and low noise (low noise figure) are important factors, linearity and low power combined with these factors is most desirable. The proposed post-distorter achieves these factors by an indirect learning architecture at base-band to compensate for nonlinearities. The low noise amplifier was designed in the standard 0.18um CMOS technology. It achieved a 10 dB improvement in linearity at 2.45GHz.

TA8a3-8

A Normalized Least Mean Fourth Algorithm with Improved Stability

Eweda Eweda, Ajman University of Science & Technology; Azzedine Zerguine, King Fahd University of Petroleum & Minerals

The paper presents a new normalized least mean fourth (NLMF) algorithm. The algorithm is derived through the minimization of the mean fourth normalized estimation error. The main advantage of the algorithm with respect to the available NLMF algorithms is that it remains stable as the input power of the adaptive filter increases. A stability step size bound of the proposed algorithm is derived. The step size bound depends on the weight initialization, while it does not depend on the input power of the adaptive filter. Simulation results support the analytical results of the paper.

TA8a3-9
An Approach to Stabilizing the Fast Array RLS Adaptive Filter Using Homogeneous Coordinates in Projective Geometry
Todd Moon, Utah State University; Kevin Hencke, University of Maryland; Jacob Gunther, Utah State University

The fast array RLS (FARLS) adaptive filter employs a circular and hyperbolic rotation to transform from a pre-array to a post-array. The algorithm is known to be numerically unstable for finite-precision implementations. It is believed that the difficulty lies in the numerical computation of the hyperbolic rotation. We propose the use of homogeneous coordinates in projective geometry to transform the hyperbolic rotation to a circular rotation. This rotation (being circular) is numerically stable. However, the transformation to the new circular representation in the projective coordinates is still numerically difficult. Nevertheless, this method may lead to a new approach that completely circumvents the instability.

TA8a3-10
Two Product-Space Formulations for Unifying Multiple Metrics in Set-Theoretic Adaptive Filtering
Masahiro Yukawa, Niigata University; Isao Yamada, Tokyo Institute of Technology

In this paper, we present two novel approaches to the issue of exploiting multiple metrics jointly for efficient adaptive filtering. The key is the introduction of product-space formulation for taking into account multiple metrics in a single Hilbert space. The first approach is based on the Pierra's idea of reformulating the problem of finding a common point of multiple closed convex sets as a problem of finding a common point of two closed convex sets in a product space. The resultant algorithm includes, as its special case, the improved proportionate normalized least mean square (IPNLMS) algorithm with a shorter range of step size. The second approach is a slight modification of the first one along the idea of constraint-embedding, thereby yielding an algorithm that includes IPNLMS with the original range of step size. The monotone approximation properties of the two algorithms are also presented. The final version will include numerical examples to demonstrate the efficacy of the presented multi-metric strategy.

TA8a3-11
On the Relation Between Blind System Identification and Subspace Tracking and Associated Generalizations
Herbert Buchner, Karim Helwani, Berlin University of Technology

Blind system identification and subspace tracking represent two important classes of signal processing problems with a variety of applications. Although originally seemingly independent from each other, the related algorithms exhibit various commonalities. In this paper, we present a novel unified derivation of the corresponding classes of adaptation algorithms. This top-down approach both clarifies the algorithmic relations and also leads to various powerful generalizations of the algorithms. Due to the rigorous approach, we obtain important practical design rules for an efficient system design. As a practical example, the acoustic localization of multiple simultaneously active sources is considered.

Track 1 – A. Communications Systems
Session: TAb1 – Coding
Chair: ***Lang Tong,*** *Cornell University*

TA1b-1 10:15 AM
Complex Number RS Coded OFDM with Systematic Noise in the Guard Interval
Mario Huemer, Christian Hofbauer, Klagenfurt University; Johannes B. Huber, University of Erlangen-Nuremberg

Recently we presented a novel OFDM (orthogonal frequency division multiplexing) signaling concept, where the cyclic prefixes (CPs) are replaced by deterministic sequences which we call unique words (UWs). The UWs are generated by appropriately loading redundant subcarriers. By that a complex number Reed Solomon (RS) code construction is introduced which can advantageously be exploited in an LMMSE (linear minimum mean square error) receiver. The overall concept clearly outperforms CP-OFDM in frequency selective channels. In this paper we introduce a method that significantly reduces the energy of the redundant subcarrier symbols by allowing some systematic noise in the UWs. The concept features a notable performance and bandwidth efficiency gain compared to our original UW-OFDM approach.

TA1b-2 10:40 AM

Extrinsic Compensation for Cycles in Message Passing Decoders

Todd Moon, Jacob Gunther, Utah State University

It is well established that cycles in the Tanner graphs associated with error correction codes introduces biases into message passing decoding and result in poor performance as the number of cycles in the graph increases. The decoder in this paper addresses the problem of cycles by computing the extrinsic probability using the previous message sent along an edge in a cycle as prior information, then computing extrinsic information using that prior as the cycle-completing message. The paper will present results of codes of various sizes and parity check densities and extrinsic cycle-breaking variations.

TA1b-3 11:05 AM

Convergence-Optimal Quantizer Design of Distributed Contraction-based Iterative Algorithms with Quantized Message Passing

Ying Cui, Vincent K. N. Lau, Hong Kong University of Science and Technology

In this paper, we study the convergence behavior of distributed contraction-based iterative algorithms for solving fixed point problems distributively with quantized message passing. We first introduce the general algorithm and analyze its convergence. Based on the derived convergence performance, we propose two quantizer designs, namely the time invariant convergence-optimal quantizer (TICOQ) and the time varying convergence-optimal quantize (TVCOQ) to minimize the effect of the quantization error on the convergence. We also study the tradeoff between the convergence error and the message passing overhead. Motivated by applications, we apply the two designs to the iterative waterfilling algorithm of MIMO interference game.

TA1b-4 11:30 AM

On Secure Communication over a Class of Degraded Relay Networks

Amir Salimi, Joerg Kliewer, New Mexico State University

In this paper we consider a class of degraded relay networks with multiple receivers, address the capacity region for the multicast case, and consider information-theoretic security aspects. Exemplary, we study a degraded relay network, which is comprised of one source, an arbitrary number of relay nodes, and two receiver nodes in a degraded setting. It is first shown that a simple combination of superposition encoding and decode-and-forward at the relay nodes achieves the capacity of this network. We then consider a related setup where the weaker receiver node now plays the role of a wiretapper which is able to listen to all the links that are terminated at the legitimate receiver. The secrecy capacity is derived for this network.

Track 1 – A. Communications Systems

Session: TAb2 – Communications Under Doppler Spread

Chair: ***Geert Leus,*** *Delft University of Technology*

TA2b-1 10:15 AM

Tracking the Time-varying Sparsity of Channel Coefficients in Shallow Water Acoustic Communications

Ananya Sen Gupta, James Preisig, Woods Hole Oceanographic Institution

The delay-Doppler spread of the shallow water acoustic channel typically follows a time-varying, complex-valued and sparse distribution. Most sparse reconstruction techniques perform poorly in the underwater paradigm due to the time-varying and ill-conditioned nature of this estimation problem. Another challenge to tracking the underwater channel coefficients is the time-varying sparsity of their distribution. Most L1-LS minimization methods assume a pre-designed sparsity factor □ for regulating the mixture of the L1-norm of the complex coefficients and the L2- norm of the estimation error. We address this issue and propose an adaptive technique that tracks the changing levels of sparsity in the channel distribution using the prediction error as a metric. Experimental results on field data validate our conjecture that the sparsity level is indeed time-varying.

TA2b-2 10:40 AM

Block Transmission over Multi-Scale Multi-Lag Wireless Channels

Geert Leus, Tao Xu, Delft University of Technology; Urbashi Mitra, University of Southern California

We consider receiver design for multi-scale multi-lag communication channels, where the propagation paths have different scales and lags. Such models are for instance encountered in underwater acoustic communications. Most existing works study the behavior of a single isolated pulse and ignore the inter symbol interference (ISI). We will look at the case where a block

of symbols is transmitted and thus ISI is present. We will first present equalizer designs based on perfect channel knowledge for such a scenario and then propose training-based channel estimation schemes. Simulation results illustrate the behavior and performance of the presented algorithms and show improvements over existing works.

TA2b-3 11:05 AM

Approximate Message-Passing-Based Decoding for Unknown Sparse Doubly Selective Channels

Philip Schniter, Ohio State University

Recently, the approach of compressed sensing has been applied to pilot-aided estimation of sparse communication channels in order to reduce the number of pilot symbols from L to O(S log L), where L denotes the channel's degrees of freedom and S the number of non-zero channel coefficients. But, even more recently, it has been shown that reliable communication over a sparse channel requires only S pilot symbols (for any L) with a receiver that does joint decoding and sparse channel estimation. The complexity of the proposed receiver is very high, though, scaling exponentially in both S and number of code blocks, K. In this paper, we propose a much simpler approach to joint decoding and sparse channel estimation that leverages recent work in approximate message passing over dense graphs.

TA2b-4 11:30 AM

OFDM over Doppler-Distorted Channels: Fractional FFT Demodulation

Srinivas Yerramalli, University of Southern California; Milica Stojanovic, Northeastern University; Urbashi Mitra, University of Southern California

We address the problem of inter-carrier interference (ICI) equalization on time-varying varying channels, which is especially challenging for high Doppler spreads and large OFDM block sizes. We propose a frequency-domain fractionally spaced equalizer and a low-complexity recursive algorithm for adapting the equalizer weights. Simulation results illustrate the improvement over conventional symbol-spaced equalizers. The proposed method requires two FFTs instead of one, but eliminates the need for explicit channel estimation, thus offering an efficient alternative to conventional methods for ICI equalization.

Track 3 – C. Networks

Session: TAb3 – Self-Organizing Networks: Architectures, Protocols and Algorithms

Chair: ***Vasileios Pappas,*** *IBM*

TA3b-1 10:15 AM

Synchronization of Coupled Oscillators

A. Kevin Tang, Cornell University

We study networks of identical phase-coupled oscillators with arbitrary underlying connected graph. By using results from algebraic graph theory, a sufficient condition is obtained which can be used to check equilibrium stability. This condition generalizes existing results and can solve some previously unsolved cases. It also leads to the first sufficient condition on the coupling function with which the system is guaranteed to reach synchronization. By the end, we will show that heterogeneous delay can help reach in-phase synchronization.

TA3b-2 10:40 AM

A Spatial Computing Approach to Distributed Algorithms

Jacob Beal, BBN

Creating distributed applications for large, decentralized networks is challenging for traditional programming approaches, posing a growing obstacle as the number and capabilities of networked devices continue to advance. In many applications, however, the network of devices is not itself of interest. Rather, we are interested in the relationships of the devices to their surroundings and their relative positions. We may thus instead write programs for the continuous space occupied by the devices, viewing the network as a discrete approximation of that space. This "amorphous medium" approach to spatial computing leads to algorithms based on manifold geometry, which are by their nature robust, adaptive, and scalable to vast numbers of devices.

TA3b-3 11:05 AM
Fast Biologically-inspired Synchronization of Periodic Events
Prithwish Basu, BBN

The synchronization of periodic events is an important and useful primitive for many aspects of ad-hoc wireless sensor networks (WSNs), e.g., synchronizing time slot boundaries for communication. Achieving and maintaining synchronization, however, is difficult due to the limited energy sources, imprecise/drifting clocks, and packet losses that plague WSNs. A pulse-coupled oscillator synchronization model by Mirollo and Strogatz has inspired several self-organizing synchronization algorithms for ad-hoc WSNs. While these algorithms are quite robust, they require reachback algorithms to avoid collisions and are not optimal in terms of convergence speed. Our proposed algorithm RefSync achieves synchronization more rapidly with reduced communication overhead through the use of a refractory period and randomness. We give an analytical proof of convergence under ideal channel conditions and demonstrate convergence (using simulation) under realistic network conditions such as random clock drifts, propagation and transmission delays, and packet losses due to channel errors and collisions in both static and mobile networks.

TA3b-4 11:30 AM
Joint Admission Control & Interference Avoidance in Self-Organized Femtocells
Kaveh Ghaboosi, Carlos H. M. Lima, Mehdi Bennis, Centre for Wireless Communications, University of Oulu; Allen B. MacKenzie, Virginia Polytechnic Institute and State University; Matti Latva-aho, Centre for Wireless Communications, University of Oulu

In this paper, we consider a femtocell deployment scenario in which radio resources are shared among self-organized femtocells. We propose a distributed Admission Control Mechanism (ACM) for traffic load balancing among sub-carriers when there are multiple Quality of Service (QoS) classes. Furthermore, we propose a mechanism based on Reinforcement Learning (RL) for slot allocation to the traffic streams on different sub-carriers, which is employed by each Femto Access Point (FAP) to mitigate interference among femtocells and the underlaid macrocell. Through simulations, the performance of the proposed schemes will be evaluated and compared to conventional Radio Resource Management (RRM) schemes.

Track 6 – F. Biomedical Signal and Image Processing
Session: TAb4 – Mathematical Methods for Biomedical Signals and Images
Chair: ***Murray Loew,*** *George Washington University*

TA4b-1 10:15 AM
Statistically Optimal Modular Partitioning of Directed Graphs
Yu-Teng Chang, Dimitrios Pantazis, Richard Leahy, University of Southern California

Network models can be used to represent interacting subsystems in the brain or other biological systems. These subsystems can be identified by partitioning a graph representation of the network into highly connected modules. In this paper we describe a modularity-based partitioning method based on a Gaussian model of a directed graph. Using the degrees of each node we first compute the conditional expected value of the connection weights. The resulting adjacency matrix forms a null model for the network which does not favor any particular partition. By comparing this null model to the true adjacency graph, we can perform a statistically optimal partitioning that maximizes modularity. As with many other partitioning methods, the solution is found using spectral matrix decomposition; in our case this matrix is the difference between the actual and null adjacency graphs. The process can be repeated to find multiple subgraphs. We demonstrate this approach through simulations and application to standard biological and other network data.

TA4b-2 10:40 AM
A Hierarchical Morphological Match Metric for Neuron Image Data
Saurav Basu, Barry Condron, Scott T. Acton, University of Virginia

An important part in our understanding of the brain is a detailed knowledge of the structure and morphology of neurons. Recently, the biology community has exerted a considerable effort in building neuronal databases that can be used for detailed modeling of the brain connectivity and relationship between cellular structure and brain function. The modeling of the brain connectivity and function requires a complete classification of neurons in the neuron database. Classifying neurons can be performed in a two step process; the first step requires the automatic segmentation or tracing of single neurons from cluttered microscopy images and the generation of neuronal morphology, and the second step requires a morphological comparison metric for neurons to cluster them into morphologically similar classes. In our approach, the first step is accomplished via an automated neuron tracing algorithm called Tree2Tree. In our solution to the comparison step, the morphology generated by Tree2Tree is employed in a match metric. Our algorithm uses a biologically inspired hierarchical model of the neuron and explores the morphology generated by Tree2Tree in order to fit the model to the segmented neuron with associated transformation parameters.

Once the associated transformation parameters from the model to the real neuron has been established, we use a hierarchical weight vector to balance the different transformations along the hierarchical model of the neuron, with higher weights assigned to model features found closer to the model root. A composite match metric is then defined as the distance between the neurons in the weighed transformation space. We have tested our algorithm on a database of drosophila ventral nerve cord neurons.

TA4b-3 11:05 AM

Fisher Information for EMCCD Imaging with Application to Single Molecule Microscopy

Jerry Chao, University of Texas at Dallas; Elizabeth Ward, University of Texas Southwestern Medical Center at Dallas; Raimund Ober, University of Texas at Dallas

Owing to its high quantum efficiency, the charge-coupled device (CCD) is an important imaging tool employed in biological applications such as single molecule microscopy. Under extremely low light conditions, however, a CCD is generally unsuitable because its readout noise can easily overwhelm the weak signal. Instead, an electron-multiplying charge-coupled device (EMCCD), which stochastically amplifies the acquired signal to drown out the readout noise, can be used. We have previously proposed a framework for calculating the Fisher information, and hence the Cramer-Rao lower bound, for estimating parameters (e.g., single molecule location) from the images produced by an optical microscope. Here, we develop the theory that is needed for deriving, within this framework, performance measures pertaining to the estimation of parameters from an EMCCD image. Our results allow the comparison of a CCD and an EMCCD in terms of the best accuracy with which parameters can be estimated from their acquired images.

TA4b-4 11:30 AM

On Parameter Estimation for Diffusion Processes in Real-time Biosensors

Manohar Shamaiah, Xiaohu Shen, Haris Vikalo, University of Texas at Austin

Molecular detection in real-time affinity-based biosensors relies on temporal sampling of the binding process in which target molecules are captured by their respective probes. The capturing process is inherently random, and is readily modeled by a stochastic differential equation (SDE). In this paper, we show that when the number of target molecules is much smaller than the number of probe molecules, the binding reaction can be described by the Cox-Ingersoll-Ross (CIR) process. Therefore, determining the number of target molecules requires finding parameters of a temporally sampled CIR process. For this, we rely on a particle filter. The transition density of the CIR process, needed for the parameter estimation, is available in closed form. For general SDE biosensor models, the transition density is approximated using Hermite polynomial expansion. Simulation results demonstrate effectiveness of the proposed method, while the full paper will present details of the particle filtering algorithm and experimental verification of the proposed technique.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TAb6 – Array Processing and Beamforming

Chair: ***Ivars Kirsteins,*** *Naval Undersea Warfare Center*

TA6b-1 10:15 AM

Efficient Frequency Invariant Beamforming using Virtual Arrays

Piya Pal, P. P. Vaidyanathan, California Institute Of Technology

In wideband array processing, frequency invariant beamforming provides a popular means to make the beampattern allpass with respect to frequency. Traditionally, such beampatterns are realized as a two dimensional filter, using tapped delay-line (TDL) filters following each spatial sensor. However it has been recently shown that with the help of a rectangular antenna array, it is possible to generate fixed frequency invariant beampatterns without using filters. In this paper, this concept is generalized to the case of two dimensional arrays with elements on a (possibly nonseparable) lattice. Since performance of the frequency invariant beamformer depends on the number of sensors which could be large for a 2D array of size MxN, a novel approach to beamforming based on the difference co-array of a physical array is also proposed, which avoids use of additional physical sensors. The realization of the frequency invariant beams using second order statistics of the impinging signal with only $M+N$ physical sensors, instead of the two dimensional array of size MxN, is demonstrated. The usefulness of the proposed method is verified through computer simulation.

TA6b-2 10:40 AM

Robust Adaptive Beamforming via Estimating Steering Vector Based on Semidefinite Relaxation

Arash Khabbazibasmenj, Sergiy Vorobyov, Aboulnasr Hassanien, University of Alberta

A new robust adaptive beamforming algorithm based on estimating the steering vector of the desired signal is proposed. It is shown that the problem of estimating the steering vector by maximizing the beamformer output power can be cast as a non-convex quadratically constrained quadratic programming problem which is in general NP-hard. However, it can be solved by using the semi-definite relaxation technique. Interesting results on the rank of the solution to the relaxed problem are obtained for our steering vector estimation problem. Simulation results demonstrate the superiority of the proposed robust beamforming algorithm over the known state of the art methods.

TA6b-3 11:05 AM

Adaptive Beamforming using Distributed Antenna Arrays: Joint versus Distributed Processing

Hongya Ge, New Jersey Institute of Technology; Ivars P. Kirsteins, Naval Undersea Warfare Center; Xiaoli Wang, New Jersey Institute of Technology

This work presents our analysis and study on adaptive beamforming and combining tech- niques applied to a system consisting of distributed antenna arrays. Our analysis reveals the connection between the natural and the practical schemes for data combining for beamforming application. The important roles played by the canonical correlations (CC), which may or may not exist among multiple data sets in a distributed antenna system, are studied to justify the use of different combining techniques.

TA6b-4 11:30 AM

Remodulation of DVB–T Signals for Use in Bistatic Passive Radar

Stephen Searle, University of Melbourne; Stephen Howard, James Palmer, Defence Science & Technology Organisation

Passive Bistatic Radar employs a local communications transmitter as an illuminator. A template for matched–filtering is obtained by steering a beam at the transmission source. However this transmission is subject to sensor noise and also may contain delayed attenuated copies of the transmitted signal. Demodulation and remodulation of the signal provides a clean template, however this also removes transmitter effects, resulting in a mismatched filter. Estimation and tracking of the carrier drift allows transmitter effects to be re–introduced to the template signal and results in ambiguity surfaces of improved quality.

Track 2 – B. MIMO Communications and Signal Processing

Session: TAb8 – Cooperative and Cognitive Transmission in Multi-Antenna Networks II 10:15 AM – 11:55 AM

Chair: ***Kaibin Huang,*** *Yonsei University*

TA8b1-1

Enhanced Limited-Coordination Strategies for Multi-User MIMO Systems

Obadamilola Aluko, Purdue University; Bruno Clerckx, Samsung Advanced Institute of Technology; David J. Love, James V. Krogmeier, Purdue University

Advanced cooperation strategies in multi-user multi-input multi-output (MU-MIMO) wireless communication systems could be facilitated by knowledge of global channel state information (CSI). Obtaining global knowledge of CSI in such MU-MIMO systems, where each user knows the CSI between every pair of users, would enhance cooperation amongst users and also improve the performance of such systems. In this paper, we utilize network coding strategies to reduce the feedback overhead required to obtain limited-rate global CSI knowledge in a MU-MIMO system. We then utilize the limited-rate global CSI to enhance MU-MIMO performance through proposed limited-coordination strategies. These enhancements include limited-coordination decoding strategies to improve decoder performance at the users, and transmission strategies for MU-MIMO relay systems.

TA8b1-2

Relay Channel with Non-causal Interference Information at the Source

Kagan Bakanoglu, Elza Erkip, Polytechnic Institute of NYU

We consider a relay channel under the presence of an external interferer where the interference is non-causally available only at the source. The interference signal may have structure, for example it could come from another source communicating with its own destination. However, the external interferer is not willing to change its communication strategy and is considered to be fixed. Two approaches are possible to mitigate the interference: exploiting the structure of the interference or treating the interference as unstructured. Using these approaches, we establish lower bounds for both the discrete memoryless and Gaussian channels and discuss to the importance of exploiting the intererference structure in multi-terminal scenarios.

TA8b1-3

Distortion-Aware Link Adaptation in Cooperative MIMO Relay Networks

Ozgur Oyman, Jeffrey Foerster, Intel Labs

In wireless multimedia communications, optimizing the quality of user experience is a major design goal, which is often quantified by the end-to-end distortion between the actual multimedia source at the encoder and its reconstructed version at the decoder. Joint source-channel coding (JSCC) techniques aim to optimize codec and radio system parameters in order to minimize end-to-end distortion and yield enhanced quality of user experience when compared with separate source-channel coding techniques. With the motivation of applying JSCC-based cross-layer optimizations to PHY/MAC layer design for practical wireless systems, distortion-aware link adaptation techniques were proposed recently for point-to-point systems toward enhanced multimedia communications. In this paper, we study these techniques in the context of cooperative relay networks composed of a single source-destination pair and multiple relay terminals in the case of two-hop communications. In particular, we propose new distortion-aware link adaptation techniques toward the selection of modulation and coding schemes (MCS), cooperative space-time processing (including cooperative diversity and cooperative multiplexing) schemes and cooperative relaying protocols with the objective of minimizing end-to-end distortion, different from classical approaches that aimed for other optimizations such as maximizing spectral efficiency or goodput. Furthermore, we investigate the performance of distortion-aware link adaptation in a realistic link-level simulation (LLS) environment based on orthogonal frequency division multiplexing (OFDM) under broadband frequency-selective fading, and demonstrate their advantages over goodput-maximizing link adaptation techniques in terms of reduced end-to-end distortion and higher peak signal-to-noise ratio (PSNR).

TA8b1-4

Beamforming on the Interference MISO Interference Channel with Multi-user Decoding Capability

Zuleita Ka Ming Ho, David Gesbert, Eurecom; Eduard A. Jorswieck, Rami Mochaourab, Dresden University of Technology

This paper considers the multiple-input-single-output interference channel (MISO-IC) in which transmitters and receivers share the same time and frequency resources. Successive interference decoding (SID) has been an essential technology on multiple access channel (MAC), but little attention has been dedicated to the scenario where receivers employ SID on interference channel. On MISO-IC, transmit beamforming vectors are designed to mitigate interference at the receivers. With SID, receivers can protentially decode interference and subtract it from the received signal and then decode the intended signal which yields a higher data rate. This brings an interesting question: when should interference be decoded and when should it treated as noise? We investigate suitable transmit beamforming strategies and explore this new achievable rate region.

TA8b1-5

Multiuser MIMO in Distributed Antenna Systems

Robert W. Heath Jr., University of Texas at Austin; Tao Wu, Young Hoon Kwon, Huawei Technologies, Co. Ltd.

Distributed antenna systems (DAS) add remote radio units with one or more antennas to enhance coverage and capacity in cellular systems. DAS works by connecting the remote radio units to the base station via a high bandwidth and low latency link. This paper proposes, analyzes, and compares several downlink multiuser multiple input multiple output (MIMO) DAS strategies in terms of per-user throughput and area spectral efficiency. Zero-forcing transmit beamforming is used at the transmitter. To facilitate rapid simulation and design space exploration, approximations of the ergodic rate are proposed for each technique assuming path-loss, small-scale Rayleigh fading, and out-of-cell interference. Treating all the remote radio units as a super MIMO transmitter gives the best performance even accounting for out-of-cell interference, though gains diminish for higher numbers of active users. Simulations compare different transmission strategies.

TA8b1-6

DMT Analysis of Opportunistic Multi-relay Networks with Different Relaying Capabilities

Mohamed Abouelseoud, Aria Nosratinia, University of Texas at Dallas

Relay selection is known to simplify signaling, avoid complex synchronization schemes, and also preserve the spatial diversity provided by the total number of relays available in the network. In this paper we investigate the effect of introducing relays of different types (relaying capabilities) into the network. We analyse an opportunistic relay selection schemes of a heterogeneous network with a single source, single destination and multiple relays with the existence of a direct link. The differential effect of adding relays of different types on the DMT is analyzed. It is shown that this will help gaining the benefits of the different relaying schemes at low and high multiplexing gains.

TA8b1-7

Throughput of Low-Power Network MIMO Cellular Systems

Shi Jin, Southeast University; Matthew McKay, Hong Kong University of Science and Technology; Kai-Kit Wong, University College London; Xiqi Gao, Southeast University

This paper investigates the uplink throughput of network MIMO cellular systems at low SNR. The network is modeled according to Wyner's extended cellular model, with intracell orthogonal medium access control. We first derive analytical expressions for the two key low SNR parameters; namely, the minimum required Eb/N0 for reliable communications and the wideband slope, assuming non-cooperative or joint (cooperative) decoding at the base stations. For the non-cooperative scenario, we find that the minimum required Eb/N0 improves with increasing numbers of receive antennas, but is independent of the number of transmit antennas and the intercell factor. The wideband slope, on the other hand, improves when either the number of transmit or receive antennas is increased, and degrades with increasing intercell factor. For the cooperative scenario, the minimum required Eb/N0 and the wideband slope improve with the cooperation at the BS. Comparing to the performance of noncooperation case, we also find that collaborative reception at the BSs is able to reduce the minimum energy per bit required for reliable communication by the intercell factor, but improves the wideband slope.

TA8b1-8

Coordinated Single-Cell vs Multi-Cell Transmission with Limited-Capacity Backhaul

Nima Seifi, Mats Viberg, Chalmers University of Technology; Robert W. Heath Jr., Jun Zhang, University of Texas at Austin; Mikael Coldrey, Ericsson AB

Base station coordination is an efficient technique to transcend the limits on spectral efficiency imposed by inter-cell interference. In this paper, we compare the performance of different coordination strategies with different amount of channel state information (CSI) and data sharing among the coordinating base stations. We especially focus on the effect of limited backhaul capacity, in a two-cell network. We show that, despite the prevailing views, coordination strategies with no data and only limited CSI sharing is preferred to those with full data and CSI sharing when the backhaul capacity is relatively low and the edge SNR is high.

TA8b1-9

Decentralized Coordinated Multi-cell Beamforming for Sum Rate Maximization

Harri Pennanen, Antti Tölli, Centre for Wireless Communications, University of Oulu

A decentralized solution is proposed for the coordinated multi-cell beamforming for weighted sum rate maximization under per BS power constraints, and with limited backhaul information exchange between BSs. The optimization problem is not convex in general, thus the global optimality cannot be guaranteed. In order to obtain a decentralized approach a standard dual decomposition method is used to decouple the original one level optimization problem into two levels, i.e., several subproblems and a master problem. The master dual problem is solved locally at each BS using a subgradient method, which requires the exchange of inter-cell interference terms between BSs. Each subproblem is solved locally at each BS using geometric programming and second order cone programming.

TA8b1-10

Statistical Beamforming in Wyner Cellular Network

Rusdha Muharar, Vasanthan Raghavan, Jamie Evans, Stephen Hanly, University of Melbourne

An infinite-array linear Wyner model for a cellular network with multiple antennas at the base-station and single antennas at the user end is considered. Due to limited antenna spacings and lack of rich scattering, the vector channels between the base-station and each user are assumed to be spatially correlated. A low-complexity transceiver architecture comprising of a statistics-based linear beamformer and a simple decoding architecture that treats multi-user interference as noise is studied. The goal of the paper is to design the beamformer architecture to maximize the sum-rate achieved by all the active users in the network. It is showed that the structure of the optimal beamformer at each base-station is a dominant generalized eigenvector of the covariance matrices

of the forward link between that base-station and the intended user, and the interfering base-station and that user. The generalized eigenvector solution is a natural extension of the single-user optimal eigen-beamforming scheme to the multi-user case. In this sense, this paper extends and builds on recent works that establish the sum-rate optimality of the generalized eigenvector solution in the broadcast and interference channel set-ups.

TA8b1-11
MMSE Transceiver Design for Coordinated Base Station Systems: Distributive Algorithm

Tadilo Endeshaw, Luc Vandendorpe, Batu Chalise, University Catholique de louvain

This paper considers the joint transceiver design for multiple-input single-output (MISO) systems with coordinated base stations (BSs). We consider two problems; 1) minimizing the weighted sum of mean-square-error (MSE) with per BS antenna power constraint and 2) minimizing the total BS power with the target MSE and the per-antenna power constraints. To solve these problems, first, for fixed receivers, we propose a computationally efficient novel distributive algorithm to get the optimal BS precoders. The proposed algorithms use Lagrangian dual decomposition, modified matrix fractional minimization and an iterative method. Second, for fixed BS precoders, the receivers are updated by the minimum MSE (MMSE) criterion. These steps are repeated until convergence is achieved. All simulation results show that the solution of our proposed distributive algorithm fits to that of the centralized algorithm.

TA8b1-12
CSI Signaling for Decentralized Coordinated Beamforming in TDD Multi-cell MIMO Systems

Petri Komulainen, Antti Tölli, Markku Juntti, University of Oulu

In this paper we present a novel multi-cell TDD CSI sounding concept for terminals employing multiple antennas. The objective is to support beam coordination with minimal or no backhaul between base stations (BS). The concept combines interference-aware CSI sounding to busy signaling so that the sounding beams corresponding to the allocated downlink data streams are considered as busy beams by the neighboring BS's. We experiment the concept in conjunction with simple regularized zero-forcing precoding and greedy beam scheduling.

TA8b1-13
Outage Probability of MISO Broadcast Systems with Noisy Channel Side Information

Alon Shalev Housfater, Teng Joon Lim, University of Toronto

Transmitter precoding strategies are necessary to achieve the capacity promised in broadcast multiple input single output (MISO) systems. However, these schemes generally require perfect channel information at the transmitter. In this paper, we investigate the impact of Gaussian noise in the channel state information (CSI) of a linear zero forcing transmitter, operating in a fading MISO broadcast channel. We consider a rectangular channel with p users and n transmit antennas such that p is no larger than n. System performance is analyzed in terms of the outage probability. Using results from an earlier work, we give simple two- and three-dimensional integral formulae of the exact outage probability for an arbitrary number of users and transmit antennas. These integrals can be numerically evaluated, for an arbitrary number of users, with a fixed amount of computational effort. We give some numerical results that afford insight to the performance of broadcast channels under transmitter-side channel uncertainty.

TA8b1-14
Multi-femtocells MIMO Processing via Amplify and Forward over the Cable (AFc)

Jonathan Gambini, Umberto Spagnolini, Politecnico di Milano

The proposed femtocell architecture exploits the benefits of centralized multi-cell MIMO processing by chaining wireless and cable MIMO channels. The home base station in femtocells is replaced by an analog-device that translates the wireless bandwidth to adapt to the wireline thus acting as an amplify and forward over the cable (AFc). Communication occurs under a hybrid two-hop transmission pattern toward a centralized processing unit that chains the air MIMO arising from the multi-user wireless links and the cable MIMO channel. After the review of the benefits of AFc for femtocell systems, here we investigate the power allocation scheme of the distributed AFc relays for downlink by accounting for OFDM transmissions and system-specific power/interference constraints over the air and cable interfaces.

TA8b1-15
Predictive Limited Feedback for Cooperative Transmission
Ramya Bhagavatula, Robert W. Heath Jr., University of Texas at Austin

In frequency division duplex systems with multicell cooperation, channel state information (CSI) can be obtained at the base stations using limited feedback strategies. Memoryless quantization can degrade sum-rates in the presence of delay, even for a large amount of feedback. In this paper, CSI is fed back using predictive vector quantization, which is a memory-based technique that exploits channel temporal correlation. An upper bound on the mean loss in sum-rate due to predictive quantization is used to derive feedback bit partitions as a function of the received signal strengths and delays. It is shown that quantizing adaptively the error between the predicted and actual CSI reduces feedback requirements in cooperative cellular systems.

TA8b1-16
Throughput Analysis of MIMO Cooperative Decode-and-Forward ARQ Protocols
Ilmu Byun, KiJun Jeon, Hyangsun You, Kwang Soon Kim, Yonsei University

There have been many researches about cooperative ARQ protocols for a SISO relay channel model. In this paper, a MIMO relay channel model consisting of a source, a destination and a relay is considered. They are assumed to be half-duplex terminals with multiple antennas. We show cooperative ARQ protocols for MIMO relay channels. The relation between the throughput and the initial transmission rates of protocols is analyzed over block fading channel, which remains constant during a block but varies from one block to another.

Track 7 – G. Architecture and Implementation
Session: TAb8 – Architectures, Implementations, and Tools I 10:15 AM – 11:55 AM
Chair: ***B. Phillips,*** *University of Adelaide*

TA8b2-1
Rate-Compatible LDPC Code Decoder Using Check-Node Merging
Anton Blad, Oscar Gustafsson, Linköping University; Meng Zheng, Zesong Fei, Beijing Institute of Technology

The use of rate-compatible error correcting codes offers several advantages as compared to the use of fixed-rate codes: a smooth adaptation to the channel conditions, the possibility of incremental Hybrid ARQ schemes, as well as sharing of the encoder and decoder implementations between the codes of different rates. In this paper, the implementation of a decoder for rate-compatible LDPC codes is considered. Assuming the use of a code ensemble obtained through puncturing of a low-rate mother code, the decoder achieves significantly reduced convergence rates by merging the check node neighbours of the punctured variable nodes. The architecture uses the min-sum algorithm with serial node processing elements to efficiently handle the wide spread of node degrees that results from the merging of the check nodes.

TA8b2-2
A Scalable and Programmable Modular Queue Manager Architecture
Qi Zhang, Roger Woods, Alan Marshall, Queen's University Belfast

The constantly changing nature of network traffic means that scalable and programmable traffic management (TM) in the form of a queue manager (QM) is becoming vital for realizing quality of service (QoS) in next generation networks. Typical QM solutions are proprietary with fixed architectures and costly management of queues. In this paper, we present a new FPGA- based QM architecture which is able to efficiently manage memory resources at high speed, thereby allowing the creation of flexible, scalable TM systems that can adapt to changing network conditions.

TA8b2-3
Hardware Implementation of DBNS Recoding for ECC Processor
Thomas Chabrier, IRISA, University of Rennes; Danuta Pamula, IRISA, University of Rennes, Silesian University of Technology; Arnaud Tisserand, IRISA, CNRS

In elliptic curve cryptography (ECC), arithmetic is a key element for designing efficient and secure cryptosystems. Finite fields arithmetic units should be fast to perform numerous and various computations (additions, subtractions, multiplications, inversions in the field) on large numbers (160-600 bits). For cost reasons, arithmetic operators should also be area, memory and power efficient. Finally, for security reasons, they should not reveal internal information during physical attacks such as side channel analysis. In this work, we study FPGA implementations of various recoding schemes for secure ECC coprocessors. In ECC protocols, the main operation is the scalar multiplication [k]P where k is a large integer (160-600 bits) and P a point on the

elliptic curve. In order to prevent from side channel analysis, k should be recoded at run time. Standard recodings schemes are Non-Adjacent Forms (NAF and w-NAF) where a signed-digit representation is used. Double-Base Number System (DBNS) has been proposed to reduce the number of non-zero digits in DBNS recoded values. DBNS is a very redundant number system and it allows sparse representations of numbers. We study the implementation of DBNS recoding schemes in FPGA for secure ECC coprocessors. We implement and analyze the cost and speed of the greedy DBNS conversion and various on-line DBNS transformations operations. We compare the performance aspects of DBNS and standard recoding schemes such as NAF and w-NAF.

TA8b2-4
Temperature Aware Power Optimization for Multicore Floating-Point Units
Wei Liu, Alberto Nannarelli, Technical University of Denmark

Fused Multiply-Add units are quite popular in floating-point execution units in state-of-the-art multicore processors. It has been shown that, for division operations, using digit-recurrence units consumes much less power and energy than using Fused Multiply-Add units which are based on Newton-Raphson approximation algorithms. In addition, thermal coupling among multipliers is reduced when digit-recurrence units are placed between them. In this work, we demonstrate the impact of different layout schemes on temperature distribution in multicore floating-point units. The results show a significant reduction in peak temperature when digit-recurrence units are used and circuit layout is carefully designed. The leakage power which increases exponentially with temperature is reduced as well.

TA8b2-5
Fast, Bit-Accurate Simulation of Truncated-Matrix Multipliers and Squarers
George Walters, Pennsylvania State University Erie; Michael Schulte, AMD Research and Advanced Development Labs

Truncated-matrix multipliers and squarers offer significant reductions in area, power, and delay, at the expense of increased computational error. These tradeoffs make them an attractive choice for many signal processing systems. However, extensive bit-accurate simulation is often necessary to explore the design space effectively and chose the best parameters. This paper presents an algorithm for fast, bit-accurate simulation of truncated-matrix multipliers and squarers in software. The algorithm is applicable to most correction methods published to date, is simple to implement, and can be used to compare different correction methods fairly and accurately.

TA8b2-6
A Redundant Decimal Floating-Point Adder
Karim Yehia, Hossam A. H. Fahmy, Cairo University

Decimal floating-point addition is important for financial and business applications. There has been a growing interest in implementing decimal floating-point adders in hardware to enhance the speed of the decimal floating-point operations. In this paper, a redundant decimal floating-point adder is proposed with the ultimate objective of enhancing the speed of the decimal floating-point addition. Redundancy allows for a carry-free addition hence the addition process does not depend on the width of the operands. The results show that our design outperforms the conventional design in both the Decimal64 and the Decimal128 IEEE format.

TA8b2-7
Arithmetic Operators Based on the Binary Stored-Carry-or-Borrow Representation
Daniel Torno, Exorand Technology; Behrooz Parhami, University of California, Santa Barbara

We introduce implementations of arithmetic operators based on the binary stored-carry-or-borrow (BSCB) representation. Several BSCB arithmetic elements, including full-adder, ripple-carry adder, and carry-lookahead adder are presented, followed by detailed design of an array multiplier. In the latter design, the conventional initial AND matrix is transformed and expressed with a redundant radix-2 representation. Each line of the resulting matrix is processed by an accumulation operator with the BSCB representation. Due to a specific property of the multiplication process, this operator is simpler than a standard full-adder cell in terms of gate count, while maintaining the same propagation latency. The entire multiplier is implemented with only XOR and AND gates, thus improving its testability and reliability.

TA8b2-8

Three Engines to Solve Verification Constraints of Decimal Floating-Point Operations

Amr Sayed-Ahmed, Hossam A. H. Fahmy, Cairo University

Decimal floating-point designs require a verification process to prove that the design is in compliance with the IEEE Standard for Floating-Point Arithmetic (IEEE Std 754- 2008).Our work represents three engines, the first engine for the verification of decimal addition-subtraction operation, the second for the verification of decimal multiplication operation, and the third for decimal fused-multiply-add operation. Each engine solves constraints describing all corner cases of the operation, and generates test vectors to verify these corner cases in the tested design. The paper describes the constraints of each operation and the steps of each engine to solve these constraints.

TA8b2-9

Algorithm and Architecture for On-Line Decimal Powering Computation

Mahmoud Hassan, Tarek ElDeeb, SilMinds; Hossam A. H. Fahmy, Cairo University

An architecture for the computation of a decimal powering function is presented in this paper. The algorithm consists of a sequence of overlapped operations: 1) digit recurrence logarithm, 2) sequential multiplication, and 3) on-line antilogarithm. A correction scheme is introduced between the overlapped operations to guarantee correct on-line calculations. Execution times are estimated for decimal64 and decimal128 formats of the IEEE 754-2008 standard for floating point arithmetic.

TA8b2-10

Degrading Precision Arithmetic for Low Power Signal Processing

Massimo Petricca, Gian Carlo Cardarilli, Università degli Studi di Roma “Tor Vergata”; Alberto Nannarelli, Technical University of Denmark; Marco Re, Pietro Albicocco, Università degli Studi di Roma “Tor Vergata”

Sometimes reducing the power dissipation of resource constrained electronic systems, such as those built for deep-space probes or wearable devices is a top priority. In signal processing, it is possible to have an acceptable quality of the signal even introducing some errors. In this work, we analyze two methods to degrade the precision of arithmetic operations in DSP to save power. The first method is based on disabling the lower (least-significant) portion of the datapath by clock-gating and forcing zeros. The second method is based on lowering the supply voltage and re-designing the carry-chains in the datapath to adapt to the increased delays.

TA8b2-11

Low-Complexity Parallel Evaluation of Powers Exploiting Bit-Level Redundancy

Muhammad Abbas, Oscar Gustafsson, Anton Blad, Linköping University

In this work we investigate the problem of computing any requested set of power terms in parallel using summations trees. This problem occurs in applications like polynomial approximation, Farrow filters etc. In the proposed powers term computation technique, the partial product of each power term is initially computed independently. A redundancy check is then made in each and among all partial products matrices at bit level. The redundancy here relates to the fact that same three partial products may be present in more than one columns, and, hence, can be mapped to the same full adder. The proposed algorithm is tested for different sets of powers and wordlengths to exploit the sharing potential.

TA8b2-12

Memristor-based Arithmetic

K’Andrea Bickerstaff, Researcher; Earl Swartzlander, Jr., University of Texas at Austin

This paper describes strategies for performing arithmetic operations in memristor-based array structures. An overview of both analog and semi-analog approaches offered in the literature for addition, subtraction, and multiplication will be given. An improved design of a memristor-based multiplier will be described.

Session: TAb8 – Architectures, Implementations, and Tools II 10:15 AM – 11:55 AM

Chair: ***B. Phillips,*** *University of Adelaide*

TA8b3-1

A New Approach for TCP/IP Offload Engine Implementation in Embedded Systems

Koji Hashimoto, Vasily Moshnyaga, Fukuoka University

TCP/IP offload engine (TOE) is an essential technology to increase throughput of network connection. In this paper we present a novel approach for TOE implementation in embedded system with very stringent requirements on area and power. Our approach is based on two design optimizations. The first one deals with architectural enhancement for reducing the size of memory buffers in TOE hardware. The second one optimizes the TCP/IP data flow in order to speculatively process the TCP/IP packet headers in parallel to DMA data transfers. Experiments show that the approach is very effective. A prototype design of TOE receiver operating at a very low (25MHz) frequency can achieve 13.1Mbs throughput while requiring only 57.5K 2-input NAND logic gates for control logic.

TA8b3-2

Scalable Multi-core Sonar Beamforming with Computational Process Networks

John Bridgman, Gregory Allen, Brian L. Evans, University of Texas at Austin

The authors present a three-dimensional sonar beamforming algorithm which is an example of an extremely parallelizable algorithm. An existing implementation of this algorithm is instrumented with OpenMP to exploit multi-core computer systems. On a two-core machine, this kernel achieves a speedup of nearly two times over a non-threaded implementation. However when executed on a 16-core machine, this kernel scales much less than expected. We implement this beamformer system within the scalable framework of Computational Process Networks to achieve additional performance CPU utilization for this larger number of cores.

TA8b3-3

ASIP Data Plane Processor for Multi-Standard Interleaving and De-Interleaving

Mohit Wani, Zoran Miljanić, Predrag Spasojević, Rutgers University; Jerry Redington, Tensilica Inc.

The design of software defined radios requires flexibility in physical layer (PHY) processing. We address an efficient implementation of flexible PHY processing for Interleaving and De-Interleaving operation through Application Specific Instruction Set Processors (ASIPs). We propose a multi- standard (802.11a, 802.16) supporting Interleaver / De-Interleaver ASIP, satisfying the throughput requirements of all the data-rates in both of the standards. The modular software implementation also allows supporting future wireless standards (that use block interleaving/ de-interleaving) with variable rates. The paper presents the architecture and ISA customizations of a reconfigurable processor, with account of area overhead for the proposed scheme and techniques to reduce it.

TA8b3-4

Architecture of a Programmable SoC for Flexible Radio Processing

Onkar Sarode, Zoran Miljanic, Predrag Spasojevic, Rutgers University

We propose a programmable System-on-Chip (SoC) architecture for flexible physical layer processing. Based on a novel hardware-oriented Virtual Flow Pipelining (VFP) framework, this solution aims at striking a balance between performance (as provided by ASICs) and flexibility (as provided by SDR). Together with providing the means for supporting heterogeneous existing and evolving Radio Access Technologies (RATs), this architecture also enables virtualization and sharing of hardware resources across simultaneous diverse traffic flows. Specifically, we propose a message-passing and clustering-based organization with a pipelined VFP controller design to meet the scalability, power and performance constraints. We have successfully programmed and executed, single as well as multiple 802.11a-like OFDM traffic flows for rates of 6, 12 and 24 Mbps.

TA8b3-5
On Prediction to Dynamically Assign Heterogeneous Microprocessors to the Minimum Joint Power State to Achieve Ultra Low Power Cloud Computing

Kranthimanoj Nagothu, Brian Kelley, Jeff Prevost, University of Texas at San Antonio

Cloud computing centers are low cost, scalable, and can process large varieties of software applications. However, the total cloud system power is high since excess processors are readily available and on reserve to service on-demand applications, as well as existing processes. We describe novel concepts that can enable the introduction of Ultra Low Power Cloud Computing systems. Our new approach involves using a variety of processors, each with different power and performance capabilities. By jointly allocating tasks to processors and dynamically turning off reserve processors, we prove that massive power reductions of up to 80% can be achieved.

TA8b3-6
Parallel - Pipelined Radix-2^2 FFT Architecture for Real Valued Signals

Manohar Ayinala, Keshab Parhi, University of Minnesota

This paper presents a novel parallel-pipelined architecture for the computation of real valued fast Fourier transform (RFFT). The proposed architecture takes advantage of the redundancy of some computations with respect to complex FFT along with low multiplicative complexity of the radix-2^2 architecture. The multi-path delay commutator architecture is used to derive a novel parallel-pipelined architecture by exploiting the redundancy in the modified flow graph. The hardware requirement for the proposed parallel architecture is minimal on both dominant components: log_4(N -1) complex multipliers and N-1 complex delay elements.

TA8b3-7
Butterfly and Inverse Butterfly nets integration on Altera NIOS-II embedded processor

Gian Carlo Cardarilli, Luca Di Nunzio, Rocco Fazzolari, Marco Re, University of Rome “Tor Vergata”; Ruby Lee, Princeton University

The Instruction Set Architecure (ISA) of microprocessors is usually word oriented, so it is not optimized to perform bit level operations. A functional unit oriented to the bit manipulation could accelerate the computation increasing the microprocessor performance in terms of execution time. This work presents the experimental results of the integration between the Bit Manipulation Unit (BMU), proposed by R. Lee, and the Altera NIOS-II processor. The BMU, described in VHDL, has been integrated in the processor using the Custom Logic feature and implemented on an Altera-Stratix FPGA.

TA8b3-8
Internal Quantization in FIR Filters Implemented Using Multiple Constant Multiplications

Guifeng (Rick) Liu, Linda DeBrunner, Victor DeBrunner, Florida State University; Kenny Johansson, Airborne Hydrography AB

In this work, we propose an approach for quantizing intermediate values within an Multiple Constant Multiplication (MCM) implementation of an FIR filter. This approach is based on calculating the sensitivity of each node. Considering the error distribution after each operation provides insight into the effects of quantization.

TA8b3-9
Effect of Order on MCM Implementations of FIR Filters

Abhijit Patil, Linda DeBrunner, Florida State University

Field Programmable Gate Arrays (FPGAs) are becoming a popular choice for digital filter implementation over DSP processors for reasons of higher sampling rates and more flexibility in design. Designing an efficient digital filter can achieved using Multiple Constant Multiplications (MCM). MCM is multiplication of constants which replaces multiplications by add and shifts. Using MCM to design a filter and selecting FPGAs for its implementation results in area saving and higher speed. This paper demonstrates how MCM affects FPGA implementation of FIR filters.

TA8b3-10

Selectable Bandwidth Filter Formed from Perfect Reconstruction Polyphase Filter Bank

fred harris, San Diego State University

There are many applications for digital filters that require operator selectable bandwidths over a wide range of fractional bandwidth. Implementation considerations guide us to favor filters with fixed coefficients that are implemented with hardwired multipliers rather than with arbitrary multipliers. The technique described here uses fixed, hardwired multipliers to form a pair of M-path analysis and synthesis filters. A selectable bandwidth filter is formed by enabling or disabling the connections between the output ports of the analysis filter and the input ports of the synthesis filter.

TA8b3-11

Reconfigurable Multiple Constant Multiplication using Minimum Adder Depth

Mathias Faust, Nanyang Technological University; Oscar Gustafsson, Linköping University; Chip-Hong Chang, Nanyang Technological University

The reconfigurable multiple constant multiplication (ReMCM) problem is finding a network of shifts, additions, subtractions, and multiplexers where a single input is multiplied with one out of several sets of coefficients. In this work, we propose an algorithm for solving the ReMCM problem. Previous publications only focus on single output problems whereas the proposed algorithm solves a multiple output ReMCM problem using a adder-graph based minimal logic depth approach. It has been shown in previous work that minimum depth MCM is advantageous from a power consumption point of view.

Track 1 – A. Communications Systems

Session: TPa1 – Advances in Multihop and Distributed Wireless Transmission

Chair: ***Raymond Knopp,*** *EURECOM*

TP1a-1 1:30 PM

Structured Lattice Codes for Wireless Relay Networks

Suhas Diggavi, Ecole Polytechnique Fédérale de Lausanne / University of California, Los Angeles

Recently, it has been shown that in many wireless communication situations lattice codes play a crucial role in transmission. We will explore the role of lattice codes in wireless relay networks. In particular we show that a lattice codes along with structured mappings at the relays approximately achieves (within a constant number of bits) the Gaussian relay network capacity for arbitrary topologies. We also examine the role of lattice codes in (approximately) achieving the rate-region for a class of wireless relay-interference networks.

TP1a-2 1:55 PM

Bounds and Lattice Strategies for Faded Relay Interference Channels

Abdellatif Zaidi, Luc Vandendorpe, University Catholique de Louvain

we consider the problem of transmission over a relay version of the carbon-copying onto dirty paper. In this setup, an additive Gaussian outside interference corrupts both transmissions to the relay and to the destination; and only the source knows the interference (in a noncausal manner). Furthermore, we focus on a faded transmission, and we model the coefficients multiplying the interference as random complex-valued coefficients. We focus on a compound channel formulation, and seek to maximize the worst case performance. We establish an achievable rate based on a coding scheme that employs lattice strategies. The results reveal that, by opposition to carbon copying onto dirty paper and its root Costa's initial dirty paper coding (DPC), it may be beneficial in our setup that the informed source uses a part of its power to partially cancel the effect of the interference so that the uninformed relay benefits from this cancellation and so the source benefits in turn. The established results may be of importance for the emerging field of cooperation in presence of some cognitive radios that might be aware of some of other users' messages intended to a common receiver.

TP1a-3 2:20 PM

Low-complexity Multiple-relay Strategies for Improving Uplink Coverage in 4G Wireless Networks

Erhan Yilmaz, Raymond Knopp, Eurecom

In this work we present low-complexity coded-modulation strategies for distributed relaying in 4G wireless networks. The primary goal of these strategies is to improve coverage on the uplink while retaining high spectral efficiency through multiuser spatial-multiplexing using two or more relays between the users and the basestation. We contrast layer 2 techniques based on full decoding at relay stations and simple compression-based (quantization) techniques with QAM alphabets. Mutual-information and error-exponent analysis clearly show the benefits of distributed quantization both in the high and medium spectral efficiency regions. We further present these results in the context of evolving LTE-Advanced standardization activities, primarily by suggesting adaptations to standardized coding and retransmission mechanisms for a multiple-relay system.

TP1a-4 2:45 PM

An Industrial Perspective of Relaying

Federico Boccardi, Volker Braun, Bell Laboratories, Alcatel-Lucent

In the last years the problem of designing efficient relaying schemes attracted a lot of efforts within the research community. If from a theoretic point-of-view important steps have been made, from an industrial perspective many points are still unclear. In this paper we focus on such an industrial perspective of relaying. First, we give an update of the status of relaying in LTE-advanced, where so-called Type-I relays will be supported for coverage extension. Then, we discuss near-future extensions, where relays will also be used for throughput enhancement. Finally, we emphasize some long-term research issues, that, if solved, would make an important impact on the design of future cellular networks.

Track 2 – B. MIMO Communications and Signal Processing

Session: TPa2 – MIMO Underwater Acoustic Communications

Chair: ***Milica Stojanovic,*** *Northeastern University*

TP2a-1 1:30 PM

Precoding in MIMO Underwater Acoustic Communications

Andrew C. Singer, Erica Daly, Jun Won Choi, University of Illinois; James Preisig, Woods Hole Oceanographic Institution

In this paper, we investigate the practical application of linear turbo equalization (TEQ) to MIMO underwater acoustic communications, with a particular emphasis on highly time-varying environments. We explore both channel estimate (CE)-based minimum mean square error (MMSE) TEQ and direct-adaptive linear TEQ. We show that substantial performance gains can be achieved in harsh environments through the use of precoding transmission. Precoding is an essential component in such links, as it eliminates low-weight code words of the “inner code” over the reals created by the time-varying ISI channel. We also show that the direct-adaptive TEQ yields performance comparable to the CE-based MMSE TEQ while maintaining lower complexity. These results are demonstrated using real experiments conducted off the coast of Martha’s Vinyard, MA (“SPACE 08”). We also discuss a practical design of a multi-channel least mean square (LMS)-based TEQ and experiments show that the LMSTEQ successfully decodes data achieving up to 19:53 kbit/s at 1km of distance.

TP2a-2 1:55 PM

Progressive MIMO-OFDM Reception over Time-varying Underwater Acoustic Channels

Jianzhong Huang, Jie Huang, Shengli Zhou, Peter Willett, University of Connecticut

Multi-carrier modulation in the form of orthogonal-frequency-division-multiplexing (OFDM) has been intensively pursued for underwater acoustic (UWA) communications recently due to its ability to handle long dispersive channels. Multiple-input multiple-output (MIMO)-OFDM is an appealing techniques in UWA systems to improve the spectral efficiency. However, its performance is limited by the crosstalk/inter-antenna interference (IAI) between antennas as well as the inter-carrier interference (ICI) due to the rapidly time-varying UWA channels. In this paper, we propose a progressive receiver for MIMO-OFDM systems, which addresses the ICI and IAI jointly to improve the system performance, based on the turbo principle. While the two-dimensional equalization has already investigated, our method is novel in that the system model for channel estimation and data detection is itself continually updated during the iterations. When the decoding in the current iteration is not successful, the receiver increases the span of the ICI and IAI in the system model and utilizes the currently available soft information from the decoder to assist the next iteration. This is in contrast to existing iterative receivers where, rather than adapting the system model to channel conditions, the system model is fixed during the iterations. Numerical simulation and experimental data collected from the SPACE08 experiment show that the MIMO-OFDM system performance can be improved significantly if the IAI and ICI are considered explicitly, and the proposed receiver can self adapt to channel variations, enjoying low complexity in good channel conditions while maintaining excellent performance in tough channel conditions.

TP2a-3 2:20 PM

Rate Bounds for Underwater Relay Channels using MIMO methods

Chiranjib Choudhuri, Urbashi Mitra, University of Southern California

The capacity of the symbol-asynchronous single-relay channel with multipath is investigated. Due to the slow propagation speed of sound in water and surface/floor reflections, such channels are highly relevant to underwater acoustic communication systems. The symbol-asynchronous single-relay channel can be modeled as a MIMO relay channel with memory; thus MIMO capacity results can be invoked to determine rate bounds. Lower and upper bounds on the capacity are derived under the assumptions of both known and unknown time delays. Conditions for characterizing capacity (when the upper and lower bounds match) are also derived and the optimality of the OFDM type of signaling can be readily inferred from the analysis. Numerical examples for practical underwater acoustic channels are provided.

TP2a-4 2:45 PM

MIMO-OFDM Receiver Design for Channels with Path-Specific Doppler Distortion

Kai Tu, Tolga M. Duman, Arizona State University; John Proakis, University of California; Milica Stojanovic, Northeastern University

Multiple input multiple output (MIMO) transmissions based on orthogonal frequency division multiplexing (OFDM) have gained significant attention for shallow water underwater acoustic (UWA) communications, thanks to OFDM's well-known advantages over intersymbol interference (ISI) channels. However, presence of Doppler scaling/shift/spread in UWA communications causes significant time variations and intercarrier interference; thus, it poses significant challenges for the implementation of MIMO-OFDM schemes. In this work, we consider these transmission systems over UWA channels where distinct propagation paths see significantly different Doppler scaling factors. Such a scenario may be motivated by cooperative communications where the transmitting nodes move in different directions with respect to the receiving element(s), hence observe very different Doppler scaling factors. As MIMO transmission protocols, we consider both the spatial multiplexing paradigm and space-frequency coding, Specifically, we determine the sufficient statistics for optimal processing at the receiver as an extension of our recent results on single-input single-output transmissions for UWA channels with path-specific Doppler. We also develop suitable receiver structures, study their performance through detailed simulations, and compare their performance with those of standard algorithms.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TPa3 – Non-Stationary Processing of Environments

Chair: ***Cornel Ioana,*** *Grenoble Institute of Technology, GIPSA-lab*

TP3a-1 1:30 PM

Stable Scatterers Detection and Tracking in Heterogeneous Clutter by Repeat-Pass SAR Interferometry

Gabriel Vasile, National Council for Scientific Research; Jean-Philippe Ovarlez, French Aerospace Lab; Frédéric Pascal, Supelec; Michel Gay, National Council for Scientific Research; Guy d'Urso, Didier Boldo, EDF

This paper presents a new estimation scheme for optimally deriving clutter parameters with high resolution repeat-pass SAR interferometry. The heterogeneous clutter in InSAR data is described by the Spherically Invariant Random Vectors model. Three parameters are introduced for the high resolution InSAR data clutter: the span, the normalized texture and the speckle normalized covariance matrix. The asymptotic distribution of the novel span estimator is investigated. A novel strategy for Stable Scatterers detection and tracking by repeat-pass SAR interferometry is also introduced by coupling sub-band / sub-aperture decomposition prior to the GLRT-LQ detector. The proposed method is tested with spaceborne InSAR images provided by the TerraSAR-X satellite.

TP3a-2 1:55 PM

Non-stationary Signal Analysis in Water Pipes Monitoring

Cornel Ioana, Grenoble INP

This paper points out on the problems related to the water pipes monitoring in the transient states. The chanhes of the hydraulic parameters creates transient over-pressures that must be carrefully analyzed. For this reason, non-stationary signal analysis tools constitutes the best candidate for this problem. This paper will illustrate the use of dynamic time warping and of the wavelet transform for the analysis of pressure of the waater pipes.

TP3a-3 2:20 PM

Non-stationary Damage State Estimation in Complex Structures Using Time Delay Embedding

Clyde Coelho, Subhasish Mohanty, Antonia Papandreou-Suppappola, Aditi Chattopadhyay, Arizona State University

In real-life scenarios, structural systems undergo fatigue loading which causes discontinuities in the material and changes in the structures strain distribution. To monitor these changes piezoelectric transducers and strain gauges will be used. The change in correlation pattern between the sensors can be mapped as a time-varying transfer function. The time-varying transfer function

can be a measure of non-stationary damage state propagation of the structure. In real-life structures fatigue damage condition can be monitored in real-time by acquiring real-time signals from the sensors. To estimate the time-series damage states, the overall fatigue life can be divided into multiple short terms discrete instances. The strain measurements at those short term discrete instances can be used to estimate the corresponding damage states. The present paper will present a time delay embedding technique to estimate the non-stationary damage states at different instances.

TP3a-4 2:45 PM

Estimation of Thermo-hydrodynamic Parameters in Energy Production Systems Using Non-stationary Signal Processing

Florin Birleanu, GIPSA-lab

Estimating the thermic and hydrodynamic parameters is a very important operation in any energy production system. Using non-invasive technique became a hot topic while it allows the estimation of the parameters without modifying the hydrolic structure of the system. In such case, the signal processing tools must be able to provide high accuracy in estimating the parameters, knowing that the signals carying the information related to this parameters are highly non-stationary. In this paper, the polynomial phase modeling is illustrated as a central tool for the estimation of temerature and flow speed of the water.

Track 6 – F. Biomedical Signal and Image Processing

Session: TPa4 – Modeling for Biomedical Imaging

Chair: ***Scott Acton,*** *University of Virginia*

TP4a-1 1:30 PM

Image-based Dynamical Modeling in Developmental Plant Biology

Amit Roy-Chowdhury, University of California, Riverside

Many biological processes are characterized by inherent, underlying dynamical patterns, e.g., the dynamics in developmental biology. Computational tools are necessary to efficiently analyze the large volumes of data being collected in the form of time-lapse images of these processes. In this presentation, we will discuss our work on modeling of dynamical processes in developmental plant biology. Pattern formation in developmental fields involves precise spatial arrangement of different cell types in a dynamic landscape wherein cells exhibit a variety of behaviors such as cell division, cell expansion and cell migration. In order to study these processes, we require computational methods for estimating the cell lineages, their volumes and expansion rates, learning their variability within and across species, and understanding the relationships between local and global growth patterns. We will present our results on tracking cell lineages, computing the cell volumes and modeling of the growth processes on the model plant Arabidopsis thaliana.

TP4a-2 1:55 PM

A 3D Cellular Resolution Gene Expression Atlas for Drosophila Embryogenesis

David Knowles, LBNL

To fully understand animal transcription networks, it is essential to accurately measure the spatial and temporal expression patterns of transcription factors and their targets. As part of a larger effort by the Berkeley Drosophila Transcription Network Project, we developed image analysis and registration methods that take image-based data from thousands of Drosophila blastoderm embryos, each co-stained for a reference gene and one of a set of genes of interest, and builds a model VirtualEmbryo. This model captures in a common framework the average expression patterns for many genes in spite of significant variation in morphology and expression between individual embryos. The method is sufficiently accurate that relationships between a pair of genes' expression inferred from the model are nearly identical to those measured in embryos co-stained for the pair. Using regression analysis methods we have shown that known gene regulatory interactions can be automatically recovered from this dataset and predict hundreds of new interactions. A database, public web site, and a user friendly visualization tool have been to developed to make the data accessible, including to those biologists with limited programming skills. We are now using these data to determine how the complex overlapping patterns of transcription factor binding we have discovered in vivo drive distinct mRNA expression patterns on each target gene, and we are developing methods to extend our expression atlas to later stages of embryogenesis.

TP4a-3 2:20 PM

Fluorescence Microscopic Imaging and Image Analysis of the Cytoskeleton

Gerlind Herberich, Thomas Wuerflinger, Antonio Sechi, RWTH Aachen University; Reinhard Windoffer, Rudolf Leube, University Hospital Aachen; Til Aach, RWTH Aachen University

Cell stability and motility depends on a complex dynamic cytoplasmic scaffolding called the cytoskeleton. It is composed of actin filaments, intermediate filaments and microtubules, and interacts with focal adhesions - multimolecular complexes responsible for the transmission of mechanical force and regulatory signals. The dynamic behaviour of these sub-cellular structures in living cells can be analysed by fluorescence microscopy yielding series of 2D or 3D images. Towards a quantitative analysis, we present methods for the segmentation and motion estimation of cytoskeletal filaments as well as for the tracking of focal adhesions, allowing the quantification of cytoskeleton dynamics under different conditions.

TP4a-4 2:45 PM

Point-Spread Function Model for Fluorescence Macroscopy Imaging

Praveen Pankajakshan, Institut Pasteur; Zvi Kam, Weizmann Institute; Josiane Zérubia, Laure Blanc-Feraud, INRIA; Gilbert Engler, INRA; Alain Dieterlen, Université de Haute-Alsace; Jean-Christophe Olivo-Marin, Institut Pasteur

Fluorescent macroscopes have been recently developed that allow the observation of relatively large samples (up to a couple of centimeters), to acquire data in three dimensions and to also perform time-lapse imaging. However, one of the main shortcoming of fluorescence macroscopy is that the observed images are affected by field aberrations. In this case, the point-spread function (PSF) varies within the lateral field and is proportional to the distance from the center of the field of view. This is because, the zoom system in a macroscope cannot achieve the condition of lateral invariance for all magnifications. A computational approach to compensate the aberration often relies on an accurate model/acquisition of the PSF. We model the PSF using the scalar diffraction approach. The pupil function is modeled by chopping it, vignetting, as a result of two limiting optical apertures not brought together to the same conjugated plane. We compare our developed model with that obtained experimentally and show the validity of our hypothesis.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TPa5 – Statistical Signal Processing for Neural Signals

Chair: ***Selin Aviyente,*** *Michigan State University*

TP5a-1 1:30 PM

Brain Controlled Robotic Platform Using Steady State Visual Evoked Potentials Acquired by EEG

Saumitra Dasgupta, Michael Fanton, Jonathan Pham, Michael Willard, Bahram Shafai, Deniz Erdogmus, Northeastern University

Abstract A noninvasive brain computer interface (BCI) based on the steady state visual evoked potential (SSVEP) has been developed and utilized in controlling an iRobot platform remotely in real-time closed-loop fashion using video feedback from the robot's eye view to the operator over the internet. The operator selects commands by focusing gaze on one of four flickering checkerboards surrounding the video feedback window in order to navigate the robot as desired. The intended/desired control commands are sent to a laptop controlling the iRobot platform via remote wireless connection. Naïve subjects are able to control and navigate the robot via the designed interface with minimal practice and classifier calibration.

TP5a-2 1:55 PM

Information-Theoretic Approaches to Identifying Parsimonious Causal Network Models of Functional Connectivity in Ensemble Neural Recordings

Christopher Quinn, Todd Coleman, Negar Kiyavash, University of Illinois at Urbana-Champaign; Nicho Hatsopoulos, University of Chicago

Improvements in electrode recording technologies have enabled neuroscience researchers to simultaneously record individual neural spike trains from large numbers of neurons. A number of researchers have attempted to identify causal relationships between simultaneously recorded neurons, though often using methods which are not robust. Recently, an information theoretic quantity, known as directed information, has been proposed as a robust measure of statistically causal relationships between neurons. A consistent estimation procedure for identifying all of the statistically causal relationships between neurons in a network has recently been proposed. However, if there is a large amount of local interactions between the recorded neurons, then there could be many statistically causal relationships between neurons. In graphical depictions of the neurons as nodes and directed arrows between neurons where there is statistically causal influence, there will be many arrows. In such cases, it could

be difficult to determine which are the most important connections. We propose to identify the key underlying structure of the network by using two provably good methods to simplify the network structure. First, we reduce the graph into a tree structure, where the remaining arrows are the most informative ones over all possible trees, using the causality equivalent of the Chow-Liu approach. Another method is to cluster neurons based on their functional connectivity, by identifying ^Óequivalence classes^Ô of neurons in the network, who have similar input connections and similar output connections. We apply this methodology on the simultaneous recordings of neurons in the primary motor cortex of an awake-behaving monkey.

TP5a-3 2:20 PM

Multi-block PLS Model for Group Corticomuscular Activity Analysis in Parkinson Disease

Joyce Chiang, Z. Jane Wang, University of British Columbia; Martin J. McKeown, Pacific Parkinsons Research Centre, University of British Columbia

Partial least square (PLS) has been proposed as means to investigate high dimensional data such as that seen in the electroencephalogram (EEG). However, in disease states such as Parkinson's Disease (PD), increased intersubject variability makes the standard PLS technique of simply pooling data from different subjects problematic. We propose a novel hierarchical two-level multiblock PLS (mbPLS) model to perform multimodal group analysis, where each subject's activation is captured in individual data blocks, and then an aggregate group-level "consensus" is obtained. We applied the mbPLS model to simultaneous EEG and electromyogram (EMG) data collected from seven normal subjects and six subjects with Parkinson's disease performing a dynamic motor task. The technique revealed task-related beta (~30Hz) activity in the motor area and determined consistent group differences in spatial patterns between normal and PD.

TP5a-4 2:45 PM

Information Theoretic Approach to Quantifying Causal Neural Interactions from EEG

Ying Liu, Michigan State University; Edward Bernat, Florida State University; Selin Aviyente, Michigan State University

In neurophysiology, it is important to quantify the causal neural interactions from electroencephalogram (EEG) data. Existing methods such as Granger causality depend on a linear model and thus cannot quantify nonlinear dependencies. In this paper, we propose to quantify the causality of the interactions using the directed information (DI) measure. A network inference algorithm is also proposed to infer the functional networks underlying EEG activity using information theoretic criteria.

Track 7 – G. Architecture and Implementation

Session: TPa6 – Computer Arithmetic II

Chair: ***N. Burgess,*** *Bristol University*

TP6a-1 1:30 PM

Instruction Set Extensions and Hardware Designs for Triple DES Processing on a Multithreaded Software Defined Radio Platform

Chris Jenkins, University of Wisconsin-Madison; Michael Schulte, AMD Research and Advanced Development Labs; John Glossner, Sandbridge Technologies

Software-defined radio (SDR) is an emerging technology that facilitates having multiple wireless communication protocols on one device. Current wireless communication protocols have significant performance requirements on this class of device. For these wireless protocols, encryption is often used to protect user data being transmitted over the air. The Triple Data Encryption Standard (DES) is an important cryptographic algorithm in wireless communication systems, and acceleration of this algorithm on SDR platforms is necessary for future generation speeds. This paper presents instruction set architecture extensions and hardware designs for Triple DES implemented on a multi-threaded SDR platform, and investigates their impact on area, performance, and power. Our results indicate that our proposed extensions and hardware designs improve the performance of Triple DES by a factor of 36 and reduce energy by 96%, while increasing average power by only 35%.

TP6a-2 1:55 PM

Opportunities for Estimating Arithmetic in Decimation Filters

Chao Liu, University of Adelaide; Oscar Gustafsson, Linköping University; Brian Ng, Braden Phillips, University of Adelaide

Estimating arithmetic gains implementation efficiency at the cost of occasional arithmetic errors. This paper evaluates opportunities for estimating arithmetic, especially estimating multipliers, in decimation filters for communication systems. The nature of the noise arising from the estimations is examined and its effect on application performance is measured. This is weighed against the benefits won in power, area and speed.

TP6a-3 2:20 PM

Computer Arithmetic Implemented with QCA: A Progress Report

Earl Swartzlander, Jr., University of Texas at Austin; Heumpil Cho, Qualcomm Inc.; Inwook Kong, Samsung; Seong-Wan Kim, University of Texas at Austin

This paper presents a review of designs for performing arithmetic operations with Quantum-Dot Cellular Automata (QCA). An overview of implementations of adders, multipliers and dividers is provided. Ripple carry, carry lookahead and conditional sum adder designs are compared. The rather surprising result is that ripple carry adders are faster in QCA than carry lookahead and conditional sum adders. Array and column compression (Wallace and Dadda) multipliers and quasi-modular multipliers are compared. The quasi-modular multipliers combine 4 N/2 by N/2 multipliers to make an N by N multiplier. Finally Goldschmidt dividers are implemented. All of the designs have been simulated with QCA Designer, a simulator that simplifies the estimation of area and delay.

TP6a-4 2:45 PM

Overcoming Double-rounding Errors Under IEEE 754-2008 Using Software

David Lutz, ARM; Neil Burgess, University of Bristol

The IEEE 754-2008 floating-point standard requires rounding to all available formats (e.g., single and double precision) from any combination of operand formats. Under the original standard, operands and results had the same format, and current hardware is likely to provide only this much. When trying to fulfill the new requirement on such hardware, there are potential double rounding problems. The problems include both incorrect incrementation and incorrect truncation. We present a software solution to both problems that requires no additional hardware, and yet still has acceptable performance.

Track 8 – H. Speech, Image and Video Processing

Session: TPa7 – Microphone Array Processing for Speech Applications I

Co-Chairs: ***Bhiksha Raj,*** *Carnegie Mellon University* ***and John McDonough,*** *Disney Research*

TP7a-1 1:30 PM

Sparse Sensing with Coprime Arrays

P. P. Vaidyanathan, Piya Pal, California Institute of Technology

Coprime sampling has, in the past, been used for signal processing applications such as range and doppler improvement in radar, and for identifying sinusoids in noise. This paper extends this idea in a number of directions and develops a sound theoretical foundation for the same. First, two uniform linear arrays with M and N elements in space are considered. With the interelement spacings of the two arrays related by coprime integers, it is shown that the number of freedoms available for DOA estimation and beamforming is O(MN). It is also shown that the difference coarray has a similar number of freedoms. This implies in particular that a passive array can identify O(MN) sources using only $M+N$ sensor elements. It is then shown that by using an M-band DFT filter bank and an N-band DFT filter bank in conjunction with the two arrays, it is possible to create a virtual filter bank with MN bands (i.e., MN beams). The increased number of freedoms can be exploited provided there is sufficient time-domain averaging capacity available. Extensions of these ideas to the case of time domain sparse sampling and two dimensional spatial sparse arrays are also outlined.

TP7a-2 1:55 PM

A Second-Order-Statistics-Based Solution for Online Multichannel Noise Tracking and Reduction

Mehrez Souden, INRS; Jingdong Chen, Wevoice Inc.; Jacob Benesty, Sofiene Affes, INRS

We propose a practical approach to multichannel noise tracking and reduction. We combine the multichannel speech presence probability (MC-SPP) that we proposed in an earlier contribution with an alternative formulation of the minima-controlled recursive averaging (MCRA) technique that we generalize from the single-channel to the multichannel case. To demonstrate the effectiveness of the proposed MC-SPP and multichannel noise estimator, we integrate them into variants of the multichannel noise-reduction Wiener filter.

TP7a-3 2:20 PM

Blind Speech Extraction Combining ICA-based Noise Estimation and Less-Musical-Noise Nonlinear Post Processing

Hiroshi Saruwatari, Yu Takahashi, Kiyohiro Shikano, Nara Institute of Science and Technology; Kazunobu Kondo, Yamaha Corp.

In this paper, we propose a new blind speech extraction microphone array combining an independent component analysis (ICA)-based noise estimator and nonlinear signal processing for achieving high-quality speech enhancement. The proposed method consists of three parts, namely, the ICA-based noise estimator for a robust target cancellation, channel-wise spectral subtraction (chSS), and post-beamforming to sum up the chSS outputs. We provide a detailed proof of the less musical noise generation property in the proposed method via higher-order statistics analysis, compared with the conventional multichannel speech enhancement methods. The superiority of the proposed method is assessed in the experimental subjective evaluation.

TP7a-4 2:45 PM

Maximum Negentropy Beamforming using Complex Generalized Gaussian Distribution Model

Kenichi Kumatani, Barbara Rauch, Saarland University; John McDonough, Disney Research Pittsburgh; Dietrich Klakow, Saarland University

This paper presents a new beamforming method for distant speech recognition. In this paper, we also investigate the beamformer for real-time operation. The beamformer proposed here is constructed in generalized sidelobe canceller configuration in order to maintain a distortionless constraint for a look direction. In contrast to conventional beamforming techniques, our beamformer adjusts the active weight vectors so as to make the distribution of beamformer's outputs as non-Gaussian as possible. That is achieved by maximizing negentropy of the outputs. The beamformer based on the maximum negentropy criterion does not suffer from the signal cancellation seen in the conventional methods. In our previous work, the generalized Gaussian probability density function (GG-PDF) for real-valued random variables (RVs) was used for modeling magnitude of a speech signal and a subband component was not directly modeled. Accordingly, it could not represent the distribution of the subband signal faithfully. In this work, we use the GG-PDF for complex RVs in order to model subband components directly. We also present a normalization method of the active weight vector which is important in real-time operation for avoiding the large active weight vector. The appropriate amount of data for adapting the active weight vector is also investigated. The performance of the beamforming techniques is investigated through a series of automatic speech recognition experiments on the Multi-Channel Wall Street Journal Audio Visual Corpus (MC-WSJ-AV). The data was recorded with real sensors in a real meeting room, and hence contains noise from computers, fans, and other apparatus in the room. The test data is neither artificially convolved with measured impulse responses nor unrealistically mixed with separately recorded noise.

Track 3 – C. Networks

Session: TPa8 – Detection & Estimation in Networks 1:30 PM – 3:10 PM

Chair: ***John Walsh,*** *Drexel University*

TP8a2-1

Delay Constrained Detection in Wireless Sensor Networks

Srikanth Hariharan, Ohio State University; Leonardo Bachega, Purdue University; Ness Shroff, Ohio State University; Charles Bouman, Purdue University

In this paper, we combine techniques to decorrelate sensor observations, and to gather data in wireless sensor networks in order to perform delay efficient signal detection. Sensor nodes measure a signal, and signals observed by different sensors could be correlated with each other. We provide a distributed algorithm that uses a Sparse Matrix Transform to decorrelate the signals. At each source node, we then compute a log-likelihood ratio of whether a signal has been detected or not. We then propose a distributed scheduling algorithm that sends the sum of the log-likelihood ratios to a sink within a deadline, under interference constraints. The sink can then compare this sum to a threshold, and decide whether a signal has been detected.

TP8a2-2

Malicious Node Detection via Physical Layer Data

Tyler Hardy, Richard Martin, Ryan Thomas, Air Force Institute of Technology

There are many mechanisms that can cause inadequate or unreliable information in sensor networks. Sensors may withhold or falsify information to obtain a disproportionate amount of resources or to disrupt network performance, physical layer phenomena may degrade the quality of the information or break a network link, and individual devices may be malfunctioning.

There are many network-layer based trust mechanisms in the literature to assess these issues; in contrast,this work develops physical-layer based trust metrics to detect malicious and malfunctioning nodes. The context will be "wireless network discovery," which refers to modelling all layers of a non-cooperative wireless network.

TP8a2-3
Secure Distributed Detection in the Presence of Eavesdroppers
V Sriram Siddhardh (Sid) Nadendla, Hao Chen, Pramod K Varshney, Syracuse University

We investigate the problem of distributed detection in the presence of eavesdroppers (Eve) in the asymptotic regime (number of sensors tending to infinity) for binary hypotheses. In the case of Eve with noisier channels, we develop our results for a general channel model and then illustrate them using two examples, BSC and AWGN channel models. We conjecture, based on our simulation results, that the LRT rule is always optimal both in terms of robustness and error probability. The above argument is corroborated with an example using BSC channels for the Eve and ideal channels for the FC. In the case of Eve with better channels, we investigate the problem of distributed detection and prove that this is a highly unfavorable scenario for the fusion center (FC). We propose a jamming scheme for the FC against Eve and evaluate the optimal distribution for the Gaussian jamming signal that requires minimum energy to make both FC and Eve's channel similar in performance.

TP8a2-4
Coding Perspectives for Collaborative Estimation Over Networks
Sivagnanasundaram Ramanan, John Walsh, Drexel University

A collaborative distributed estimation problem over a communication constrained network is considered from an information theory perspective. A suitable architecture for the codes for this multiterminal information theory problem is determined under source-channel separation. In particular, distributed source codes in which each node multicasts a different message to each subset of other nodes are studied. This code construction hybridizes multiple description codes and codes for the CEO problem. The goal of this paper is to determine the fundamental relationship between the multicast communication rates and estimation performance obtainable. An achievable rate distortion region is proved to this problem and its structural properties are studied. Also, this achievable rate region is shown to simplify to the known bounds to some simpler problems.

TP8a2-5
Distributed State and Field Estimation Using a Particle Filter
Florian Xaver, Christoph Mecklenbräuker, Vienna University of Technology; Peter Gerstoft, University of California, San Diego; Gerald Matz, Vienna University of Technology

This paper addresses decentralized tracking of the probability distribution of state parameters of a space-time-variant process described by a linear partial differential equation using a particle filter (PF). We focus on localizing an acoustic source in a given region. The underlying wave equation leads to a high-dimensional estimation problem. We propose a technique for reducing the computational effort of the PF which allows to distribute the signal processing algorithm over the nodes of a wireless sensor network (WSN) without a fusion center.

TP8a2-7
Distributed Gauss-Newton Method for Localization in Ad-hoc Networks
Benjamín Béjar Haro, Pavle Belanovic, Santiago Zazo Bello, Universidad Politécnica de Madrid

Energy efficiency, scalability and robustness are key features of Ad-hoc and Wireless Sensor Networks and the use of decentralized algorithms is of practical importance in such scenarios. A method for node localization is proposed by solving a nonlinear least-squares problem in a distributed fashion. For that purpose we propose a Gauss-Newton algorithm with embedded consensus that requires only local communication and converges to the centralized version.

TP8a2-8
Multitarget Tracking with the Cubature Kalman Probability Hypothesis Density Filter
Davide Macagnano, Giuseppe Thadeu Freitas de Abreu, Centre for Wireless Communications, University of Oulu

In this paper we investigate the problem of jointly estimating a time varying number of targets and their locations from sets of noisy range measurements received at fixed anchor nodes with known location in presence of association uncertainty. To do so we use a recent generalization of the Bayesian approach to the multitarget problem that goes under the name of Probability Hypothesis Density (PHD) recursion, which admits a closed-form solution as a Gaussian Mixture (GM). Because of the nonlinearity between observations and state model existing in the formulation of the GM-PHD filter, we solve the multivariate integral occurring in the nonlinear Bayesian formulation employing the new Cubature Kalman Filter (CKF). The performance for the proposed CKF-GM-PHD filter is compared against its linearized version of the filter. The results show that the CKF-based solution is far more robust than the EKF-based solution both in terms of cardinality as well as in terms of location estimates.

Track 2 – B. MIMO Communications and Signal Processing

Session: TPa8 – Low Complexity Implementation and Receiver Issues 1:30 PM – 3:10 PM

Chair: ***Raghu Rao,*** *Xilinx*

TP8a1-1
A Low Complexity Square Root MMSE MIMO Decoder

Raghu Rao, Helen Tarn, Raied Mazahreh, Chris Dick, Xilinx, Inc.

As MIMO technology proliferates in present day wireless communication systems, efficient architectures for MIMO Decoding are being pursued. In the last few years, an alternate formulation of the MMSE MIMO Decoder for spatial multiplexing MIMO systems called the square root MMSE MIMO Decoder based on QR decomposition, has appeared in the literature. This approach presents better fixed precision performance and reduces the number of multiplications required for the MMSE solution. It avoids inverting the upper triangular matrix but at the cost of dealing with a larger matrix, the extended channel matrix. In this paper, we present an approach based on the square root MMSE scheme where we don't pay the price for the extended channel matrix in terms of increased hardware resources but at the same time benefit from its better fixed precision properties thereby achieving a solution with much lower complexity. We present a scalable, systolic array based solution where the systolic array scales with the number of transmit antennas and not with the number of receive antennas. Hence, the complexity of the MIMO Decoder does not increase even as the number of antennas at the receiver (base-station) is increased.

TP8a1-2
Low-Complexity Seysen's Algorithm based Lattice Reduction-Aided MIMO Detection for Hardware Implementations

Lukas Bruderer, Christian Senning, Andreas Burg, ETH Zurich

Lattice reduction aided linear detectors for MIMO systems are a promising receiver structure for low complexity implementations. We present a modified Seysen's algorithm tailored for hardware implementation. The calculation of the Seysen's algorithm based on the Gramian matrices alone reduces both, the number of arithmetic operations and the storage requirements. A novel threshold-based early termination scheme provides a trade-off between average run-time and implementation loss. Furthermore, we show that limiting the maximum number of iterations in order to guarantee a minimal processing rate only slightly reduce the improvements of LR, provided that a greedy selection procedure is used to find the best update coefficient.

TP8a1-3
Low Complexity PARAFAC Ceceiver for MIMO-OFDMA System in the Presence of Multi-Access Interference

Avik Santra, Hari K.V.S., Indian Institute of Science

Synchronization errors in time and frequency occur in OFDM, OFDMA leading to interferences that complicate data detection at the receiver. The trilinear data model for MIMO-OFDMA for any arbitrary carrier assignment scheme is formulated. We propose to accomplish user's separation at the BS using estimates of the multiple CFOs and then use PARAFAC-TALS for blind estimation of user's parameters. We show that the proposed receiver not only relaxes the identifiability criteria but also simplifies the tri-linear alternating least squares iteration.

TP8a1-4
Adaptive Stream Mapping Multi Antenna Systems with Low Complexity Iterative Detection

Danshan Chen, Alister Burr, University of York

In this paper, we propose an adaptive linear precoding technique combined with low complexity iterative detection which can significant improve the performance of spatial multiplexing MIMO systems. Previous work had been done on selection criterion to select the appropriate precoding matrix to perform the stream mapping, based on ML or MMSE detection. However these selection criterion were designed under the uncoded conditions. In this paper we present an selection criterion based on the Extrinsic information transfer (EXIT) chart specifically designed for a reduced complexity iteratively detected coded system

with interference cancellation detection. The results show that the detector maintains full diversity and has performance within 0.4 dB of the ML detector. Moreover precoding increases performance by 1.2 dB compared to the ML detector with fixed stream mapping. And it also beats the SNR criterion from previous work.

TP8a1-5
A Unified Receiver for MIMO Communication With Imperfect Channel Knowledge
Meriam Rezk, Benjamin Friedlander, University of California, Santa Cruz

The problem of MIMO communications with imperfect channel knowledge at the receiver is considered. Channel estimation errors are due to noise and due to time variations induced by Doppler. A receiver structure named the Unified Generalized Likelihood Ratio Detector (UGLRD) is presented. The UGLRD is based on joint symbol-channel estimation, and it accommodates variations in the availability and the reliability of the channel state information at the receiver. Hence, it can also be applied to a blind or to a differential MIMO system. The performance of this receiver is compared with that of the conventional MIMO receiver for different channel conditions. It is shown that the UGLRD outperforms the conventional receiver at the expense of higher computational complexity.

TP8a1-6
Performance of a MIMO Receiver Using Joint Channel-Symbol Estimation in the Presence of Channel Errors
Meriam Rezk, Benjamin Friedlander, University of California, Santa Cruz

This paper analyzes the performance of a MIMO receiver which uses joint channel-symbol estimation. The analysis accounts for situations where no channel information is available at the receiver, as well as situations where imperfect channel estimate is available. The former case corresponds to blind MIMO. For the latter case, channel estimation errors due to noise and due to time variations induced by Doppler effects are considered. A closed form solution is derived for the block pairwise error probability at high SNR. The result involves taking the expectation of the Q-function which is evaluated using Guass-Chebychev quadrature numerical integration rule. Based on the obtained pairwise error probability, the total bit error rate of the system is predicted. The accuracy of the results is verified through computer simulations accounting for different channel scenarios and different variations in the reliability of the channel state information at the receiver.

TP8a1-7
Design of High Performance MIMO Receivers for LTE/LTE-A Uplink
Meilong Jiang, Narayan Prasad, NEC Labs America, Inc.; Xiaodong Wang, Columbia University

In this paper, we investigate high performance multiple-input-multiple-output (MIMO) receivers for the DFT-Spread-OFDM based long term evolution (LTE) cellular uplink, wherein multiple users are scheduled on the same resource block via space division multiple access (DFT-Spread-OFDM-SDMA). The proposed receiver schemes will also apply to the LTE-Advanced cellular uplink where simultaneous transmission of multiple streams by a single user will be allowed. Two types of advanced non-linear receivers are considered and optimized, namely, a receiver based on a two-symbol soft-output demodulator (referred to as the two-symbol MLD receiver) and a turbo minimum mean squared error successive interference cancelation (turbo MMSE-SIC) receiver. Based on extensive simulations, it is shown that both the two-symbol MLD and the turbo MMSE-SIC receivers exhibit superior performance compared to the conventional linear MMSE (LMMSE) receiver. In general, turbo MMSE-SIC receiver offers the best performance but also introduces larger latency and higher computational complexity. The proposed two-symbol MLD receiver with new pairing method has a moderate complexity and a performance that lies between those of the turbo MMSE-SIC and LMMSE receivers.

TP8a1-8
Generalized Spatial Modulation
Abdelhamid Younis, University of Edinburgh; Raed Mesleh, Jacobs University Bremen; Harald Haas, University of Edinburgh

In this paper, a generalized technique for spatial modulation (SM) is presented. Generalized spatial modulation (GSM) overcomes the constraint in the number of transmit antennas needed by SM in a novel fashion. In GSM, a block of information bits is mapped to a constellation symbol and a spatial symbol. The spatial symbol is a combination of transmit antennas activated at each instance to transmit the constellation symbol. This is unlike SM where single transmit antenna is activated at each instance. Hence, GSM increases the overall spectral efficiency by base-two logarithm of the number of antenna combinations, thus reducing the number of transmit antennas needed. The performance of GSM is analysed in this paper, and an upper bound on the bit-error-ratio (BER) performance is derived. Also, an algorithm to optimise the antenna combination selection is proposed. Finally, the performance of GSM is validated through Monte Carlo simulations and compared to the performance of SM where it is shown that GSM performs nearly the same as SM, with a significant reduction in the number of transmit antennas.

TP8a1-9
Decision Directed Channel Estimation for Reducing Pilot Overhead in LTE-A
Johanna Ketonen, Markku Juntti, University of Oulu; Jari Ylioinas, Nokia Siemens Networks

The use of decision directed (DD) channel estimation in a MIMO-OFDM downlink receiver with a LTE pilot structure is studied in this paper. The space-alternating generalized expectation-maximization (SAGE) algorithm is used to improve the receiver performance from the least-squares (LS) channel estimator. The benefit of using DD channel estimation can be observed with higher user velocities, where channel estimation from the pilot symbols gives a poor performance. The DD channel estimation can also be used to reduce the pilot overhead. The complexity of the SAGE channel estimator is also discussed.

TP8a1-10
A Novel Structure for MMSE Transceivers over Slowly Time-varying Channels
Chih-Hao Liu, P. P. Vaidyanathan, California Institute of Technology

This paper addresses the design problem of decision feedback (DF) transceiver without zero-forcing constraint over slowly time-varying narrowband multi-input multi-output (MIMO) channels. The space-time generalized triangular decomposition (ST-GTD) is applied for the design of minimum mean square error (MMSE) DF transceiver. With space-time powerloading, the proposed space-time geometric mean decomposition (ST-GMD) MMSE transceiver maximizes Gaussian mutual information over the equivalent channel seen by each space-time block. For practical applications, the causal ST-GTD MMSE transceiver which does not require channel prediction but shares the same asymptotic bit error rate (BER) performance with the ST-GMD MMSE system is also developed. In high signal to interference plus noise ratio (SINR) region, our results show that the proposed MMSE transceivers have better BER performance than the conventional GMD-based MMSE transceiver; the average BERs of the proposed systems are a non-increasing function of the ST-block size.

TP8a1-12
Multi-User Beamforming and User Pairing For WiMAX
Thomas Svantesson, Pengcheng Zhan, Gokhan Korkmaz, ArrayComm, LLC

WiMAX is a broadband internet access standard for both fixed and mobile wireless networks. In WiMAX, base stations (BSs) with beamforming capability do not announce beamforming weights explicitly. Instead, weights are applied to pilot subcarriers in units of major groups. Shorter packets such as voice over IP (VoIP) packets do not span a full major group creating the problem of multiple users in the same major group using the same beamforming weights. This paper considers calculating those multi-user beamforming weights and pairing users in a practical WiMAX BS. A criterion for pairing users and a tree-search algorithm to find the optimal pairing is proposed with a complexity suitable for implementation. Furthermore, a criterion for the multi-user weights is presented as well as methods of calculating the weights. Finally, performance results from a WiMAX VoIP scenario show that the performance when paired multiple users use the same weights only is 0.5dB off single user beamforming results and providing a large gain over a standard cyclic-delay-diversity (CDD) beamforming strategy.

Track 5 – E. Array Processing and Statistical Signal Processing
Session: TPa8 – Techniques in Networking and Communications 1:30 PM – 3:10 PM
Chair: ***A. Lee Swindlehurst,*** *University of California, Irvine*

TP8a3-1
Optimal MISO Pre-Equalization for Filter Bank Based Multicarrier Systems
Marius Caus, Ana Isabel Pérez-Neira, Technical University of Catalonia (UPC)

One of the aspects that degrades multicarrier modulation (MCM) systems performance are the spectral nulls that multipath channel may present. As a result, the affected subcarrier signals may be severely degraded resulting in a loss of symbols. In order to combat this effect we have studied how to overcome this problem by taking advantage of spatial diversity along with the knowledge of channel state information in transmission (CSIT). In particular we propose the design of a per-subcarrier broadband beamformer for filter bank multicarrier (FBMC) systems. Simulation-based results demonstrate the efficacy of the proposed optimal and suboptimal designs in saving power given QoS constraints.

TP8a3-2
Mean Shift Based Segmentation for Time Frequency Analysis of Packet Based Radio Signals

Goran Ivković, Predrag Spasojević, Ivan Šeškar, Rutgers University

We consider the problem of RF signal analysis where one sensing node observes a frequency band possibly used by multiple packet based radio transmitters producing signals with non-persistent excitation. Since each combination of active transmitted signals results in a composite signal with its distinct statistical properties, the received signal at the sensing node consists of a number of statistically homogeneous segments. First important task in the analysis of this type of signals is to localize these segments in time. We propose a segmentation algorithm for solving this problem. Initial segmentation is obtained using a variant of mean shift algorithm with adaptive scale parameters. We provide a convergence analysis of this algorithm and propose a method for selecting scale parameters. Final estimates of change points are obtained by iteratively optimizing certain criterion function. Proposed algorithm is almost completely nonparametric and it finds the number of segments automatically. Performance of the algorithm is studied using simulations involving signals used in 802.11 networks. The algorithm can be applied in a scenario where the sensing node is part of a radio scene analysis network providing information that can be used for achieving efficient utilization of radio spectrum.

TP8a3-3
An AOA Estimator for Multiple GPS Signals Using a Modified Despreader

Suk-seung Hwang, Chosun University; John Shynk, University of California, Santa Barbara

The Global Positioning System (GPS) is one of the core technologies of a Location-Based Service (LBS), and has numerous applications, from the leisure business to the military. LBS requires accurate location information of an object, which GPS provides by using at least four satellites. Angle-of-arrival (AOA) information of the GPS satellites can be used to enhance the received signal quality, but this can be difficult to estimate because the signals operate with a low signal-to-noise ratio (SNR) and in the presence of high-power interference. In this paper, we explore an extension of a recently proposed GPS AOA estimation technique based on a modified despreader for multiple GPS signals. Since the extended estimator does not require a singular value decomposition (SVD), unlike a conventional AOA estimation algorithm such as Multiple Signal Classification (MUSIC), it has a relatively low computational complexity. Also, this technique does not need to isolate the GPS AOAs from other signals because it does not estimate the AOAs of the interference. We demonstrate the performance of the GPS AOA estimation algorithm for multiple satellites via a computer simulation example.

TP8a3-4
Distributed Source Coding in Large Wireless Sensor Networks

Joan Enric Barcelo Llado, Antoni Morell Pérez, Gonzalo Seco Granados, Universitat Autonoma de Barcelona

We study the performance improvement of \emph{distributed source coding} in large wireless sensor networks using enhanced correlation estimators. Distributed source coding is especially useful when data correlation exists since they try to remove the redundancy in the information. Particularly, dense sensor networks are rich in correlations. Some existing results from information theory show that this compression can be executed in a distributed fashion and without any loss of performance in comparison with the centralized approach. However, there is still a gap in performance between the theoretical bound and the results achieved with practical implementations. To mitigate this, we propose to use enhanced correlation estimators. Simulation results suggest a performance improvement in mean square error and symbol error rate in favor of classical methods.

TP8a3-5
An Active Distributed Approach for Cyber Attack Detection

Hoa Nguyen, Sandeep Gutta, Qi Cheng, Oklahoma State University

With fast growing cyber activities everyday, cyber attack has become a critical issue over the last decade. A number of cyber attack detection algorithms have been developed and applied in this field of study with different levels of success. In this paper, a new distributed cyber attack detection algorithm based on the decision cost minimization strategy is introduced. The proposed algorithm employs sensor selection and active training techniques to reduce computational complexity for real time implementation without decreasing its effectiveness. The algorithm includes a data fusion rule to combine the decisions from distributed local binary classifiers using the decision cost function. KDD 1999 datasets are used to evaluate the proposed method. It is shown that the proposed detection system is a more flexible and suitable cyber attack detection solution for both known and unknown cyber attacks.

TP8a3-6

Distributed Signature Learning and Calibration for Large-Scale Sensor Networks

Naveen Ramakrishnan, Emre Ertin, Randolph Moses, Ohio State University

In this paper, we consider the problem of joint sensor calibration and target signature estimation using distributed measurements from a large-scale wireless sensor network with random link variations. Specifically, we propose a new Distributed Space-Alternating Generalized Expectation Maximization (EM) algorithm, DSAGE, which can estimate the (constrained) parameters of interest, using measurements from the sensor nodes, in a distributed manner. Unlike a centralized algorithm that relies on pooling measurement vectors from the network, DSAGE operates at the parameter space reducing the communication bandwidth. We model the sensor network as a connected graph and show that the gossip-based distributed consensus can be used to update the estimates at each iteration of the DSAGE algorithm. As a result the proposed algorithm is robust to link and node failures, unlike previously suggested distributed subgradient methods that rely on formation and maintenance of a stable network infrastructure to perform iterations in parameter space. We prove the guaranteed convergence of the algorithm to the centralized data pooling solution and compare its performance with the derived Cram\'{e}r-Rao bound, using simulations.

Track 1 – A. Communications Systems

Session: TPb1 – Wireless Communications

Chair: ***Aydin Sezgin,*** *University of Ulm*

TP1b-1 3:30 PM

Identifying Wireless Users via Power Amplifier Imperfections

Sepideh Dolatshahi, Georgia Institute of Technology; Adam Polak, Dennis Goeckel, University of Massachusetts Amherst

Variations in the RF chain of radio transmitters can be used as a signature to uniquely associate wireless devices with their transmissions. In this work algorithms for deciding about the origin of a given message of interest are developed based on differences in characteristics of RF power amplifiers (PAs) modeled with Volterra series. In particular, for a two users scenario, a generalized likelihood ratio test (GLRT) and a classical likelihood ratio test are considered. The corresponding decision rules and performance expressions are derived. Finally, the applicability of the proposed approach is shown through measurements performed on PAs used in today's WLANs.

TP1b-2 3:55 PM

Full-duplex Wireless Communications Using Off-the-Shelf Radios: Feasibility and First Results

Melissa Duarte, Ashutosh Sabharwal, Rice University

We study full-duplex wireless communication systems where same band simultaneous bidirectional communication is achieved via cancellation of the self interfering signal. We present an analysis of the self interference cancellation mechanism and derive expressions for the average power of the self interfering signal after RF and baseband cancellation. We present experimental results that show that the equations we have derived are a good fit for the measured data and the average percentage error is less than 4.1% for all the scenarios considered. Perfect self interference cancellation is not possible in real systems due to noise in the estimates of the signals required for cancellation. We show that self interference can be suppressed sufficiently such that the resulting full-duplex system capacity lies between the capacity of a half-duplex Multiple Input Multiple Output (MIMO) system using diversity techniques and the capacity of a half-duplex spatial multiplexing MIMO system. This result opens the possibility for the design of new systems and network protocols based on FD radios.

TP1b-3 4:20 PM

Low Complexity Approximate Maximum Throughput Scheduling for LTE

Stefan Schwarz, Christian Mehlführer, Markus Rupp, Vienna University of Technology

In this paper we address the challenge of multiuser scheduling in the downlink of 3GPP UMTS/LTE. Long Term Evolution (LTE) imposes the constraint of using the same code rate, modulation order and transmit power for all resources a User Equipment (UE) is scheduled onto. This, in addition to the lack of channel knowledge, prohibits theoretical concepts such as capacity maximization to be applied for resource allocation. Based on the Channel Quality Indicator (CQI) feedback we derive a linearized model for multiuser scheduling. In contrast to other proposals we use Mutual Information Effective SNR Mapping (MIESM) to calculate an average CQI value for all UE resources. This enables a rate increase while still guaranteeing an imposed Block Error Ratio (BLER) constraint. The proposed framework can also be applied to implement other scheduling strategies. This is demonstrated by comparing different standard schedulers in terms of achieved throughput and fairness.

TP1b-4 4:45 PM

A Stochastic Association Mechanism for Macro-to-Femtocell Handover

Carlos H. M. Lima, Kaveh Ghaboosi, Mehdi Bennis, Centre for Wireless Communications, University of Oulu; Allen B. MacKenzie, Virginia Polytechnic Institute and State University; Matti Latva-aho, Centre for Wireless Communications, University of Oulu

The femtocell paradigm will enable next generation wireless networks by enhancing indoor coverage, increasing capacity, and offloading macrocell traffic. In this paper, we consider a macrocell site underlaid with self-organized stand-alone femtocells and propose a mechanism for macro-to-femtocell handover, inspired by multi-stage Dutch auction. The proposed solution introduces a new distributed handover procedure, which does not rely on any centralized coordination typically performed by the Macro Base Station (MBS) in existent cellular systems. Through simulations, the performance of the proposed scheme will be evaluated, showing its efficiency in performing seamless handover to serve Macro Users (MUs).

Track 2 – B. MIMO Communications and Signal Processing

Session: TPb2 – MIMO for Ad Hoc Networks

Chair: ***Nihar Jindal,*** *University of Minnesota*

TP2b-1 3:30 PM

Transmission Capacity of Multi-antenna Ad Hoc Networks with CSMA

Jeffrey Andrews, Radha Krishna Ganti, Andrew Hunter, University of Texas at Austin

Multiple antennas have become a common component of wireless networks, improving range, throughput, and spatial reuse, both at the link and network levels. At the same time, carrier sensing is a widely used method of improving spatial reuse in distributed wireless networks, especially when there is limited coordination among non-communicating nodes. While the combination of carrier sensing and multiple antennas has been considered in the literature, physical layer spatial models and the attendant consequences have not been included. The primary reason for this has been the difficulty of analyzing functionals of interacting point processes. Having developed new methods of quantifying physical layer performance with robust spatial network models, we use these techniques to address the following questions: What multiple antenna techniques produce the best network performance, and what is the performance gain? And, how should multiple antennas interact with carrier sensing parameters? Overall, the analysis confirms the significant benefit of multiple antennas in distributed wireless networks.

TP2b-2 3:55 PM

MIMO Beamforming with Quantized Feedback in Ad Hoc Networks: Transmission Capacity Analysis

Matthew McKay, Hong Kong University of Science and Technology

We investigate the performance of MIMO beamforming with quantized feedback in ad hoc networks. The primary findings are that a moderate number of feedback bits are sufficient to obtain significant transmission capacity gains compared to non-feedback schemes, whilst also achieving a high percentage of the transmission capacity obtained with unlimited feedback. We demonstrate that this achievable percentage is larger in high path loss environments, and that the number of feedback bits to maintain a fixed gain increases with the number of transmit antennas. These findings are obtained by new transmission capacity expressions which we derive.

TP2b-3 4:20 PM

Optimal SISO and MIMO Spectral Efficiency to Minimize Hidden-Node Network Interference

Daniel Bliss, Lincoln Labs

In this letter, the optimal spectral efficiency for a given message size that minimizes the probability of causing disruptive interference for ad hoc wireless networks or cognitive radios is investigated. Implicitly, the trade being optimized is between longer transmit duration and wider bandwidth versus higher transmit power. Both single-input single-output (SISO) and multiple-input multiple-output (MIMO) links are considered. Here, a link optimizes its spectral efficiency to be a “good neighbor.” The probability of interference is characterized by the probability that the signal power received by a hidden node in a wireless network exceeds some threshold. The optimization is a function of the transmitter-to-hiddennode channel exponent. It is shown that for typical channel exponents a spectral efficiency of slightly greater than 1 b/s/Hz per antenna is optimal.

TP2b-4 4:45 PM

Optimized Multi-Antenna Communication in Ad Hoc Networks with Opportunistic Routing

Nihar Jindal, Niranjay Ravindran, Peng Wu, Joseph Blomer, University of Minnesota

We consider MIMO ad hoc systems with an opportunistic routing scheme that allows for selection of the best immediately available relay. For single stream transmission, when the transmission probability is chosen optimally, the progress-rate-density (which is proportional to end-to-end rate), increases linearly for increasing number of receiver antennas, even when each transmitter has only a single antenna. When the transmitters are equipped with multiple antennas, we optimize the number of spatial streams per node to maximize the progress-rate-density.

TP2b-5 5:10 PM

The Role of Channel Distribution Information in the Cross-Layer Design of Opportunistic Scheduler for MIMO Networks

Sheu-Sheu Tan, University of California, San Diego; Adam Anderson, University of South Florida; James R. Zeidler, University of California, San Diego

In this paper, we design a new class of cross-layer opportunistic channel access scheduling framework for multiple-input multiple-output (MIMO) networks through exploitation of channel distribution information (CDI) in a form of full spatial correlation matrix. Due to propagation delay, infrequent channel updates, and time-varying node position, channel knowledge becomes outdated and scheduler based solely on channel state information (CSI) can experience severe performance degradation. By utilizing the higher stability of channel statistics such as CDI, we can achieve `differential' time-scale equalization in MIMO networks between Physical (PHY) and Media Access Control (MAC) layer and reduce the frequency and amount of feedback between nodes.

Track 3 – C. Networks

Session: TPb3 – Network Information Theory

Chair: ***Hamid Sadjadpour,*** *University of California, Santa Cruz*

TP3b-1 3:30 PM

Opportunistic Interference Alignment effects in Cooperative Broadcast of Multiple-Source

Saeed Bagheri, University of California, Davis; Shrut Kirti, Cornell University; Anna Scaglione, University of California, Davis

In this paper we are concerned with a scenario in which multiple messages (M) are concurrently flooding a dense wireless network. Rather than using standard packet switching we assume that the nodes relay cooperatively the messages, if they are able to decode them. Our objective is to propose and analyze a scheme that exploits the alignment of interference that opportunistically arises in the relay process. More specifically, each receiver observes the M messages over N dimensions through an MIMO random channel. For each message, there exists an MIMO channel matrix that is associated with the interference. The key idea is to use the spatial occurrence of MIMO interference matrices that have a high condition number to design good beamformers to decode the messages. We refer to this effect as opportunistic interference alignment. We show that this scheme outperforms other cooperative schemes in terms of packet delivery ratio.

TP3b-2 3:55 PM

Study of Throughput and Latency in Finite-buffer Coded Networks

Nima Torabkhani, Georgia Institute of Technology; Badri Vellambi Ravisankar, University of South Australia; Faramarz Fekri, Georgia Institute of Technology

The exact queueing analysis of erasure networks with network coding in a finite buffer regime is an extremely hard problem due to the large number of states in the network. In such networks, packets are lost due to either link erasures or blocking by the full buffers. In this paper, a block-by-block random linear network coding scheme with feedback on the links is selected for reliability and more importantly guaranteed decoding of each block. We will propose a novel method that iteratively estimates the performance parameters of the network and more importantly reduces the computational complexity compared to the exact analysis. The proposed framework yields a fairly accurate estimate of the probability distribution of buffer occupancies at the intermediate nodes using which we can obtain analytical expressions for network throughput and delay distribution of a block of packets.

TP3b-3 4:20 PM

Asymptotic Interference Alignment for Network Coding Applications

Viveck Cadambe, Syed Jafar, Hamed Maleki, University of California, Irvine

The strengths and limitations of the asymptotic interference alignment scheme proposed by Cadambe and Jafar for the K user interference channel are investigated in the context of network coding applications - such as distributed data storage and harvesting.

TP3b-4 4:45 PM

Outage Analysis and Optimization for Block Asynchronous Users

Amir Khandani, University of Waterloo

This paper addresses a Gaussian interference channel consisting of two active users. The channel from each transmitter to each receiver is modeled as quasi-static and non-frequency selective Rayleigh fading. Users are asynchronous, meaning there exists a mutual delay between their transmitted codes. Due to the randomness of delay, no user is aware of the location of the interference burst along its code. As such, no interference cancellation is performed. By the same token, interference has a mixed probability density function as a result of ambiguity on the start of the interference bursts. Due to non-ergodicity of the model, outage analysis is an appropriate tool to study the system performance. All users follow a locally Randomized On-Off (ROO) signaling scheme where each transmitter quits transmitting its Gaussian signals independently from transmission to transmission with certain probability. A Lower bound on the so-called outage capacity per user is developed using entropy power inequality and a key upper bound on the differential entropy of a mixed Gaussian random variable. It is shown that by adopting the ROO scheme, the outage capacity is strictly improved compared to a scenario where both users keep transmitting their Gaussian signals at all time.

Track 4 – D. Adaptive Systems and Processing

Session: TPb4 – Adaptive Filters - Theory and Applications

Chair: ***José Carlos M. Bermudez,*** *Federal University of Santa Catarina*

TP4b-1 3:30 PM

A Stochastic Analysis of the NLMS Algorithm Implemented in Finite Precision

Neil Bershad, University of California, Irvine; Jose Carlos M. Bermudez, Federal University of Santa Catarina

Abstract—Quantization effects in the NLMS algorithm are investigated for a white Gaussian data model. Nonlinear recursions are derived for the weight mean error and mean-square deviation (MSD) that include the effects of various quantization operations in the implementation of the algorithm. The nonlinear recursion for the MSD is solved numerically and shown in excellent agreement with Monte Carlo simulations, supporting the theoretical model assumptions. The theory is extended to tracking a Markov channel and accurately predicts the tracking behavior as well. For the white data case, the excess mean square-error (EMSE) is simply related to the MSD. The tradeoff between the number of bits in the quantizers, steady-state EMSE, and algorithm convergence rate is studied using these results.

TP4b-2 3:55 PM

Comparison of LMS and NLMS Adaptive Filters with a Non-stationary Input

Eweda Eweda, Ajman University of Science & Technology

The tracking performances of the LMS and NLMS algorithms are compared when the input of the adaptive filter is nonstationary. The analysis is done in the context of tracking a Markov plant. A periodic scenario of the variation of the input power is considered. The steady-state peak mean square deviation is used as the tracking performance measure. It is found that one algorithm outperforms the other depending on the values of the rate of variation of the input power, the minimum input power, the noise variance, and the mean square plant parameter increments.

TP4b-3 4:20 PM

Steady State Analysis of the Conventional CLMS and Augmented CLMS Algorithms for Noncircular Complex Signals

Danilo Mandic, Yili Xia, Imperial College; Scott Douglas, Southern Methodist University

The recently introduced augmented complex least mean square (ACLMS) algorithm is shown to be suitable for the processing of both second order circular (proper) and noncircular (improper) signals. This is achieved by using the widely linear model, where the associated covariance and pseudocovariance matrices contain complete second order statistical information available. In theory, both the linear and widely linear model achieve the same mean square error for propers signals, whereas the widely

linear model exhibits lower mean square error for improper signals. In principle, similar conclusions can be drawn about the behaviour of the linear CLMS and widely linear ACLMS, however, due to their different dynamics of learning and possible noncircularity of the input, there is a need for a rigorous assessment of their dynamics of learning. To this end, we employ the energy conservation principle to analyse the convergence and steady state properties of ACLMS and CLMS when dealing with noncircular complex signals. Simulations in the adaptive prediction setting for signals with different degrees of noncircularity support the the analysis.

TP4b-4 4:45 PM

An Alternate View of Nonlinear Adaptive Filters

Tokunbo Ogunfunmi, Santa Clara University

Kernel adaptive filter algorithms have been introduced recently.These algorithms use the theory of reproducing Kernel Hilbert spaces to allow us to perform adaptive filtering in a linear space that is nonlinearly related to the original input space. It also helps to relate the new developments in machine learning and adaptive filtering communities. Some of the issues to be resolved when using Kernel adaptive filters include (i) need to select a kernel function (ii) need for regularization and (iii) need for curtailing the growth of the filter structure. Nonlinear systems have also been modeled using the Volterra, Weiner and Hammerstein models. For any nonlinear system, it may be very difficult to compute Volterra model coefficients/kernels. However, for some particular interconnections of linear, time-invariant (LTI) subsystem and nonlinear memory-less subsystems, it is possible. We have recently developed algorithms for adaptive filters based on Volterra, Wiener and Hammerstein models. We have applied these algorithms to real applications with good results. In this paper, we study the concept of "surprise" as introduced in [1] for kernel adaptive filters and its use in a sparsification scheme to curtail the growth of the adaptive filter. We define a similar concept as a means of characterizing nonlinear adaptive filters based on polynomial models. We relate it to the widely used concept of "innovations" in linear adaptive filters. We also explore various interconnections of adaptive parallel cascade structures for implementing these nonlinear adaptive filters.

TP4b-5 5:10 PM

Diffusion LMS with Communications Constraints

Øyvind Lunde Rørtveit, John Håkon Husøy, University of Stavanger; Ali H. Sayed, University of California, Los Angeles

Abstract—Diffusion LMS is a distributed algorithm that allows a network of nodes to solve estimation problems in a fully distributed manner by relying solely on local interactions. The algorithm consists of two steps: a consultation step whereby each node combines in a convex manner information collected from its neighbors and an adaptation step where the node updates its local estimate based on local data and on the data exchanged with the neighbors. Various forms of diffusion algorithms are possible such as combine-then-adapt (CTA) and adapt-then-combine (ATC) forms, in addition to probabilistic implementations where consultations are performed only with a subset of the neighbors chosen at random. In this paper we propose an alternative protocol to reduce the communications cost during the consultation process. Each node is limited to selecting only one of its neighbors for consultation, and we propose a dynamic technique that enables the node to pick from among its neighbors that neighbor that is likely to lead to the best mean-square deviation (MSD) performance. In other words, rather than picking nodes at random, the proposed algorithm is meant to enable nodes to perform the selection in a more informed manner. The paper describes the proposed method and illustrates its behavior via simulations.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TPb5 – Integrated Multimodal Sensing

Chair: ***Muralidhar Rangaswamy,*** *Air Force Research Laboratory*

TP5b-1 3:30 PM

Agile Multi-modal Tracking with Dependent Measurements

Jun Zhang, Arizona State University; Quan Ding, Steven Kay, University of Rhode Island; Antonia Papandreou-Suppappola, Arizona State University; Muralidhar Rangaswamy, Air Force Research Laboratory

We investigate the problem of multi-modal target detection and tracking using joint dependent radio frequency (RF) and electro-optical (EO) sensor measurements. We propose the use of a recursive Bayesian track-before-detect (TBD) particle filter that incorporates detection statistics from all RF sensor range-Doppler resolution cells and EO sensor angle resolution cells. The RF-EO measurements are assumed dependent, as expected in realistic scenarios, and with joint distribution functions with unknown parameters. As a result, we integrate exponentially embedded distribution families in the TBD formulation.

TP5b-2 3:55 PM

Sensor Integration for Classification

Steven Kay, Quan Ding, University of Rhode Island; Muralidhar Rangaswamy, Air Force Research Laboratory

In the problem of sensor integration, an important issue is to estimate the joint PDF of the measurements of sensors. However in practice, we may not have enough training data to have an accurate estimate. In this paper, we have constructed the joint PDF using an exponential family for classification. This method only requires the PDF under a reference hypothesis. Its performance has shown to be as good as the estimated maximum a posteriori classifier which requires more information. This shows a wide application of our method in classification because less information is needed than existing method.

TP5b-3 4:20 PM

Closed-Loop Tracking Using Multimodal RF/EO Sensors

Sean O'Rourke, A. Lee Swindlehurst, Center for Pervasive Communications and Computing

This paper addresses the problem of using multiple, spatially distributed RF/EO sensors for target tracking. Data from the multi-modal sensors is collected at a fusion center, and used to track multiple objects in the sensors' field of view. Continuous approximations are made of the sensors' resolution in range, range rate and azimuth in terms of system parameters such as bandwidth, PRF, EO zoom and pointing angle. These approximations are used to create a continuous objective function in terms of the tracking filter's information matrix, and a search is undertaken to find the system parameters that optimize tracking performance based on the current target environment. Simulations demonstrate the benefit of this technique compared with open-loop tracking methods.

TP5b-4 4:45 PM

Design and Performance of a Multimodal Radar Test-Bed for Progressive Resolution Enhancement

Surendra S. Bhat, Ram M. Narayanan, Pennsylvania State University

With the available EM spectrum becoming increasing scarce, a crucial requirement is one of multimodal sensor operation. This paper describes a multimodal radar system capable of providing target indication with a large range extent and progressively switching to a narrow range extent HRR mode for extracting recognizable target features. The multimodal radar consists of a test-bed that will enable the generation of linear frequency modulation waveforms of various bandwidths. A narrow bandwidth waveform is used to obtain low-range resolution profiles of the target initially. High-range resolution processing is then progressively performed using waveforms of appropriate higher bandwidths within selected range cells wherein targets are declared. This paper discusses the design of the radar and the experimental results obtained from field testing.

Track 7 – G. Architecture and Implementation

Session: TPb6 – Computer Arithmetic III

Chair: ***A. Tenca,*** *Synopsys*

TP6b-1 3:30 PM

Complex Division with an FMA

Claude-Pierre Jeannerod, INRIA, Universite de Lyon; Nicolas Louvet, Universite Claude Bernard Lyon 1, Universite de Lyon; Jean-Michel Muller Muller, CNRS, Universite de Lyon

We review classical algorithms for complex division, and we propose new "compensated" algorithms for this operation. These algorithms take advantage of the availability of a fused multiply-add (FMA) instruction to provide more accurate complex quotients at a reasonable cost. An important problem is to avoid as much as possible overflows in the intermediate computations, as well as harmful underflows

TP6b-2 3:55 PM

Shaping Probability Density Function of Quantization Noise in Fixed Point Systems

Karthick Parashar, Daniel Menard, Romuald Rocher, Olivier Sentieys, University of Rennes-1, IRISA/INRIA

Fixed point refinement of signal processing algorithms is a necessary and crucial step during implementation. Increasing system complexity and the presence un-smooth operators whose behavior cannot be analyzed with existing analytical models forces the use of simulation for evaluating the effects of fixed-point word-lengths. Mixed simulation approaches which makes use of analytical models as a means of acceleration are proposed. The idea behind mixed simulation approach is to analytical models for the arithmetic operators and uses simulation at the boundaries of un-smooth operators. Thereby avoiding simulation of the entire system and hence saving time. At the boundaries however, the analytical models must be robust enough to faithfully

represent the fixed-point effects. Given the sensitivity of un-smooth operators to their input, it is no longer sufficient to just have noise power. The faithful representation of the fixed-point phenomena requires the knowledge of additional noise characteristics such as the probability density function (PDF) and the noise spectral density. This paper proposes a model for shaping the PDF of the total quantization noise at the output of a given arithmetic operator based system. It improves upon a previous PDF model to and provides a generic frame-work which can be used for any arithmetic operator based system. The idea behind the proposed approach is to identify a cluster of noise sources such that they add up to be a Gaussian and hence reduce the computation involved in determining the noise PDF. In the proposed model, the PDF is modeled as a sum of one Gaussian distributed and a number of uniform distributed noise sources. A simple algorithm to find a cluster of noise sources which can add up to provide the best Gaussian is described. The KURTOSIS function is used to quantify the Gaussian-ness of the sum of random variables. As a representative signal processing algorithm, a basic 32 tap FIR filter is used to test the proposed algorithm and the initial results are provided. The PDF obtained by execution of algorithm followed by very little analytical computation (such as convolution of PDFs and evaluating the Central limit theorem) is seen to be closely matching with the PDF obtained by simulation. The proposed technique provides the basic approach and promising first results towards a generic model to quickly shape the noise PDF at the output of any arithmetic operator based signal processing system.

TP6b-3 4:20 PM

Towards a Highly Efficient Implementation of Sequential Montgomery Multiplication

Joao Carlos Neto, University of Sao Paulo; Alexandre Tenca, Synopsys, Inc.; Wilson Ruggiero, University of Sao Paulo

A method to generate highly efficient implementations of sequential Montgomery multipliers (MM) is proposed. It is applied to radix-2 MM but could be used for other radices. An efficient solution is obtained when inactive adders in a cycle are re-assigned to perform useful computation. The resulting hardware algorithm and architecture accelerate the modular multiplication by looking ahead the input data of two iterations and in some cases compressing two iterations in one, without significantly increasing the iteration time. Experiments show 37.8% average reduction in clock cycles when proposed multiplier is applied to implement modular exponentiation in the 2048-bit RSA cryptosystem.

TP6b-4 4:45 PM

Multi-Operand Decimal Addition by Efficient Reuse of a Binary Carry-Save Adder Tree

Alvaro Vazquez, INRIA; Elisardo Antelo, University of Santiago de Compostela

This paper presents the design of a combined binary/decimal multi-operand adder intended for high performance applications. Decimal operands are represented in a special decimal coding different than BCD, which allows the use of a binary carry-save adder to perform decimal addition with a small correction. Previous proposals implement this correction introducing additional logic in the critical path or by a rearrangement of the binary tree. This has an significant impact on the latency of the binary operation. In our case, the decimal corrections are performed in parallel with the computation of the binary sum, so that the binary tree layout is not modified.

TP6b-5 5:10 PM

On Equivalences and Fair Comparisons Among Residue Number Systems with Special Moduli

Behrooz Parhami, University of California, Santa Barbara

Properties and applications of residue number systems (RNS) with special moduli of the form 2^k plus-or-minus 1, with a single power-of-2 modulus often also included, have been studied extensively. We show that lack of systematic studies has led to rediscovery of “new” moduli sets that are really equivalent to previously studied ones and that certain comparisons presented to show advantages of some proposed moduli sets are rather unfair. We prove a general mathematical result that allows us to normalize the single power-of-2 modulus, thus removing some of the problematic variations from such proposed sets. We then offer an assessment strategy based on dynamic ranges of the RNS sets being compared, rather than on artificial parameters that may be different for comparable systems.

Session: TPb7 – Microphone Array Processing for Speech Applications II

Co-Chairs: ***Bhiksha Raj,*** *Carnegie Mellon University* ***and John McDonough,*** *Disney Research*

TP7b-1 3:30 PM

Online Meeting Recognizer with Multichannel Speaker Diarization

Shoko Araki, Takaaki Hori, Masakiyo Fujimoto, Shinji Watanabe, Takuya Yoshioka, Tomihiro Nakatani, NTT Communication Science Laboratories

We present our newly developed real-time conversation analyzer for group meetings. The goal of the system is to automatically estimate "who speaks when and what" in an online manner. In our system, "who speaks when" information is first obtained by estimating the directions of arrival (DOAs) of signals. Then, "who speaks what" is estimated with our automatic speech recognition (ASR) system, after suppressing reverberation, background noise, and interference speakers' voices. In this paper, we particularly focus on the speaker diarization ("who speaks when" estimation) method, and we show that the speaker diarization information helps the ASR to reduce insertion errors.

TP7b-2 3:55 PM

Group Delay Based Methods for Recognition of Distant Talking Speech

Rohan Mandala, Mrityunjaya Shukla, Rajesh Hegde, IIT Kanpur

The group delay function has been used conventionally in temporal spectral analysis and feature extraction for speech recognition. In this work we present a detailed analysis of a novel approach to spatial spectral analysis of speech using the MUSIC-Group delay spectrum. In our previous work we have proposed the use of the MUSIC-group delay spectrum [icassp 2010], for direction of arrival estimation (DOA) and distant speech recognition. We discuss the advantages of the proposed method in terms of resolving closely spaced speech sources with minimal number of sensors. The analogy between temporal and spatial spectral analysis of speech using the MUSIC-group delay method is described using pole zero plots. This method is also analyzed from a minimum phase perspective as is done in temporal processing of speech. Additional analysis is performed using the Pisarenko-Group delay spectrum in terms of real time performance. DOAs estimated from the proposed approach are used to train filter and sum beamformers. Distant speech recognition experiments in clean and reverberant conditions using the beamformed speech signal indicate reasonable improvements over correlation and sub space based methods.

TP7b-3 4:20 PM

Microphone Array Processing for Distance Speech Capture: A Probe Study

Tao Yu, Chi Zhang, John H.L. Hansen, CRSS: Center for Robust Speech Systems

In this study, we consider advancements in microphone array processing for speech applications where vocal effort is either variable, or low intensity. The motivation here is to develop speech signal processing strategies which are capable of tracking and processing speakers who have consistent low vocal effort due to either speech production problems, vocal pathologies, or generally speak softly or in a whisper mode. Here, we explore strategies to track variable vocal effort speakers, as well as identify these speech islands from the array processing audio stream. Experiments will be demonstrated for actual audio collected in conference room/classroom scenarios.

TP7b-4 4:45 PM

A Prototype of Distant-talking Interface for Control of Interactive TV

Maurizio Omologo, Fondazione Bruno Kessler

This paper aims to describe goals, challenges, and main achievements of DICIT EC project, which addressed the development of a multi-modal user-friendly interface for control of SetTopBox, TV and related services. The interface includes a microphone array to support distant-talking voice input with multiple active speakers. The front-end processing component feeds a chain including speech recognition, natural language understanding, and spoken dialogue management components. The resulting prototype was replicated at several sites and evaluated by 170 users. Results of this campaign showed effectiveness of the adopted solution as well as potential for future development of real smart space applications.

TP7b-5 5:10 PM

An Acoustic Front-End for Interactive TV Incorporating Multichannel Acoustic Echo Cancellation and Blind Signal Extraction

Klaus Reindl, Yuanhang Zheng, Anthony Lombard, Andreas Schwarz, Walter Kellermann, University of Erlangen-Nuremberg

In this contribution, a novel acoustic front-end for distant-talking interfaces as developed within in the European Union-funded project DICIT (Distant-talking interfaces for Control of Interactive TV) will be presented. This novel front-end comprises state-of-the-art multichannel acoustic echo cancellation and blind source separation-based signal extraction. Considering the overall system complexity, user acceptance and cost, it is of great importance to only use a small number of microphones. Therefore, in contrast to an existing concept that uses 13 microphones, the proposed front-end only uses two microphones. The proposed novel acoustic front-end will be analyzed and evaluated for different realistic scenarios.

Track 3 – C. Networks

Session: TPb8 – Scheduling, Relaying and Routing 3:30 PM – 5:10 PM

Chair: ***Phil Schniter,*** *Ohio State University*

TP8b1-1

Admission Control Based Joint Bandwidth and Power Allocation in Multi-User DF Relay Networks

Xiaowen Gong, Sergiy Vorobyov, Chintha Tellambura, University of Alberta

The problem of joint admission control and resource allocation in multi-user decode-and-forward relay networks with limited bandwidth and power resources is considered. A suboptimal greedy search algorithm is developed to solve the problem efficiently. The condition under which the proposed algorithm is optimal is derived. The performance improvements offered by the proposed method is demonstrated by simulations. The advantages of the suboptimal greedy search algorithm for admission control are also shown.

TP8b1-2

Broadcast-Relay-Broadcast Channels

Liang Chen, University of Maryland

In this paper, we study the achievable rate regions of broadcast-relay-broadcast channels. The source broadcasts information to the users. A number of relays are used to assist users. Each relay receives information from the source and forwards information to users. The relays also broadcast the forwarding information to users. Achievable rate regions for broadcast-relay-broadcast channels are derived in the discrete memoryless case. We also provide an outer bound on the rate regions, which is tighter than the cut-set bound.

TP8b1-3

Opportunistic Scheduling Using ARQ feedback in Multi-Cell Downlink

Sugumar Murugesan, Philip Schniter, Ness Shroff, Ohio State University

We study cooperative, opportunistic multiuser scheduling using ARQ feedback in multi-cell downlink systems. Adopting the cell breathing ICI control mechanism, we formulate the scheduling problem as an infinite horizon discounted reward partially observable Markov decision process and study two scenarios. When the cooperation between the cells is asymmetric, we show that the optimal scheduling policy has a greedy flavor and is simple to implement. Under symmetric cooperation, we link the scheduling problem with restless multiarmed bandit processes and propose a low complexity index scheduling policy. The proposed index policy is essentially Whittle index policy, if the scheduling problem is Whittle indexable. Extensive numerical experiments suggest that the proposed policy is near-optimal.

TP8b1-4

Routing Policy-dependent Hop Count Distribution in Wireless Ad Hoc Networks

Golaleh Rahmatollahi, Leibniz University of Hannover; Giuseppe Abreu, University of Oulu

We consider the problem of determining the prob- ability mass function (PMF) of the number of hops (a.k.a hop-count distribution) required to reach a designated distance D in two-dimensional ad hoc networks with a given hopping policy (random, closest- and furthest-neighbor). Starting with one-dimensional (linear) networks, we first employ an analytical framework whereby the hop-count distribution under different hopping policies is obtained easily using a renewal process of the hop-length distribution and the total number of nodes within D. An immediately consequence of this approach is that the difficulty

of handling the statistics of various sums of random variables is avoided, such that no recursive formulae or complicated convolutions are required. Another equally important feature is that the method is general, requiring only that the distribution of the hop-length under the desired policy be known, such that application to two-dimensional networks is straightforward, and amounts the problem of computing the projection of each 2D hop onto the straight line between source and destination. The accuracy of the resulting PMF's is confirmed via the Kullback- Leibler divergence between the analytical formulae and empirical distributions obtained from Monte-Carlo trials.

TP8b1-5

Polar Codes for Compress-and-Forward in Binary Relay Channels

Ricardo Blasco-Serrano, Ragnar Thobaben, Vishwambhar Rathi, Mikael Skoglund, Royal Institute of Technology

In this paper we construct polar codes for binary symmetric relay channels with orthogonal receiver components. We show that polar codes achieve the cut-set bound when the relay-destination link supports compress-and-forward relaying based on Slepian-Wolf coding. More generally, we show that the compress-and-forward rate is achievable using polar codes for Wyner-Ziv coding. In both cases the block error probability can be bounded as $O(2^{-N^{\beta}})$ for $0<\beta<\frac{1}{2}$ and a sufficiently large blocklength N

Track 6 – F. Biomedical Signal and Image Processing

Session: TPb8 – Biomedical Signals and Images 3:30 PM – 5:10 PM

Chair: ***Murray Loew,*** *The George Washington University*

TP8b3-1

Prediction of Biologically Active Regions in Protein Sequences via Best Basis Selection

Ravi Narasimhan, Applied Micro Circuits Corporation

A common biological function of a set of protein sequences can be identified using the resonant recognition model (RRM), where frequency analysis of the protein sequences and peak detection are used. In this paper, local amino acid regions associated with a common biological function are predicted using space-frequency analysis of protein sequences. After a relative shift of each protein sequence, the highest-amplitude space-frequency tile of the best basis is used to predict biologically active regions. Joint processing of sequences using best basis selection provides fine spatial resolution in active region prediction, which can be useful for peptide synthesis and drug design.

TP8b3-2

Combination of a FIR Filter with a Genetic Algorithm for the Extraction of a Fetal ECG

Mohamed Amine Guettouche, Malika Kedir, Assya Bousbia-Salah, University of Sciences and Technology Houari Boumediene (USTHB)

The detection and analysis of a fetal cardiac signal are the primary goals of the electronics in fetal monitoring. Adaptive Wiener filtering and quadratic error minimization such as RLMS (Recursive Least Mean Square) and NLMS (Normalized Least Mean) methods can give a satisfactory but non-optimal solution. In this paper, we propose to apply an adaptive filtering by combining a time-varying finite impulse filter (FIR) with a genetic algorithm (GA). With this solution, we can obtain the filter's coefficients which minimize the quadratic error and guarantee convergence towards the optimal filter. In order to show the impact of GA compared to the filters of Wiener, RLMS and NLMS, we realized on the same real signal recorded on mother (MECG), the extraction of the cardiac signal of fetus (FECG). An GA of eight bits and ten iterations only seems to be a filter of quality compared to the other filters.

TP8b3-3

Modeling of the Beat of a Cardiac Signal by Gaussians

Malika Kedir, Hafid Hariz, Saliha Ould-Slimane, University of Sciences and Technology Houari Boumediene (USTHB)

The goal of this work is to break up the beat of a cardiac signal ECG into a linear combination of Gaussians with various averages and various standard deviations. For that we carry out a library of 132 Gaussians with 132 different averages and standard deviations. The research of the maximum of the scalar product of each Gaussian of the library by the beat signal ECG to be modelled, permits to find Gaussian the most relevant. Our results shows that the number of Gaussian does not exceed five and that the first Gaussian found in each case, is that which corresponds to wave QRS. At initialization step, the algorithm Orthogonal Forward Regression, achieves an error of 10-4. The error is established between the real signal of MIT database ECG and the modeled signal . The projected gradient algorithm we achieved an optimization of the first order

TP8b3-4

Optimal Estimation in DNA Microarrays via Global Optimization

Sang Hyun Lee, Manohar Shamaiah, Haris Vikalo, University of Texas at Austin

DNA microarray technology relies on affinity between complementary nucleic acids to detect the presence and estimate the amounts of target molecules of interest. Molecular binding is stochastic in nature, having inherent uncertainty manifested as Poisson noise. This, along with interference due to non-specific binding, are among the main obstacles for achieving high accuracy of DNA microarrays. In this paper, optimal target estimation in DNA microarrays is shown to lead to an NP-hard fractional program. A practically feasible approach which employs a branch-and-bound algorithm solving a convex optimization problem in each step is presented. Simulation results demonstrate that the proposed approach outperforms previously considered methods.

TP8b3-5

Design and Implementation of a Long Range Iris Recognition System

Justin De Villar, Robert Ives, James Matey, US Naval Academy

Iris recognition systems are primarily limited by constraints imposed on subjects to enable the collection of images of sufficient quality for recognition. A typical distance from the eye to the sensor is approximately 1 ft. This paper introduces the Iris-at-a-Distance (IAAD) system, a prototype that has proven the feasibility of iris recognition at a distance of 30 meters. This dual camera system features an 8-inch telescope that makes collection of adequate iris images at this distance possible. This paper describes the system design and presents preliminary performance results using a commercial version of the Daugman iris recognition algorithm.

TP8b3-6

Using an FPGA to Accelerate the Hough Transform in Iris Recognition

Jennifer Shafer, Hau Ngo, Robert Ives, United States Naval Academy

Iris recognition is a topic of great interest in the information security field. The demand for fast and accurate identification is widespread. The circular Hough transform is a proven method for detecting and isolating the iris within the image of the eye. This paper implements this portion of the iris recognition algorithm on an FPGA improving performance and decreasing overall compute time.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: TPb8 – Statistical and Adaptive Signal Processing 3:30 PM – 5:10 PM

Chair: ***Victor DeBrunner,*** *Florida State University*

TP8b2-1

CDF Resampling for Dataset Expansion in Gaussian Mixture Models Density Estimation

Alessio Medda, The Henry M. Jackson Foundation for the Advancement of Military Medicine, USAARL; Victor DeBrunner, Florida State University

This paper provides a study on the performance of density estimator presented Medda and DeBrunner for short datasets, when using resampling techniques to expand the original sample set. In a novel approach that employs a goodness-of-fit measure to estimate the correct model order, the quality of the estimated mixture depends on the complexity of the true density and the length of the sample set. The poor performance experienced when estimating densities from short datasets can be corrected using a simple resampling of the empirical cumulative distribution, used to generate additional samples. This technique produces a clear improvement in the quality of the estimation and results in mixtures that better approximate the true densities.

TP8b2-2

Time Reversal Beamforming of Guided Waves in Pipes with a Single Defect

Nicholas O'Donoughue, Joel Harley, Jose' M.F. Moura, Carnegie Mellon University

Structural health monitoring of buried pipelines is an important application for aging infrastructures. Ultrasonic guided wave inspection is an attractive tool, due to the long propagation of guided waves in the wall of a hollow cylinder. However, guided waves present a unique environment with heavily multi-modal signal propagation and complex dispersion (frequency-dependent propagation speeds). In order to alleviate these challenges, conventional techniques rely on high-voltage excitation with complex transducer arrays, but these systems are not conducive to a monitoring solution. Instead, they require periodic excavation and testing. In prior work, we have shown that Time Reversal allows for reliable detection with relatively simple antenna arrays that can be operated in low-power. This paper focuses on localization of these defects. We utilize a beamforming approach that makes use of theoretical dispersion curves to generate fault images. We show through simulations that Time Reversal Beamforming achieves high-resolution localization of a fault in the presence of strong dispersion and heavily multi-modal propagation.

TP8b2-3
On the Predictability of Phase Noise Modeled as Flicker FM Plus White FM

Siamak Yousefi, Joakim Jalden, Royal Institute of Technology

In this paper, the predictability of RF oscillator phase noise (PHN) is investigated when a more accurate model than Wiener process is considered. The increment of the PHN process, i.e., the frequency fluctuation, modeled as the superposition of stationary 1/f process and white noise, is predicted by a linear-minimum-mean-square-error (LMMSE) predictor. The normalized MSE (NMSE) in one-step-ahead (OSA) prediction of PHN is then evaluated from the result obtained and is compared for different SNRs and PHN characteristics. In certain situations, the new PHN model shows a good prediction gain compared to the Wiener model, even for low order predictors.

TP8b2-4
Detection of Circular and Noncircular Signals in the Presence of Circular White Gaussian Noise

Xi-Lin Li, Tulay Adali, Matthew Anderson, University of Maryland, Baltimore County

Determination of the effective order of the signal subspace is a critical problem in many signal processing and communications applications. The formulation proposed by Wax and Kailath [1] based on information theoretic criteria has been widely used in this context, however, it assumes a circular model for the signals, which obviously is a limiting assumption for many applications. In this paper, we approach the problem as one of model selection rather than pure order selection such that besides the selection of the effective order, we also determine the number of circular and noncircular signals in the presence of circular white Gaussian noise. The method reduces to Wax and Kailath's signal detection method [1] when all signals are circular, and as we show with simulations, provides a significant performance gain over Wax and Kailath's method in the presence of noncircular signals.

TP8b2-5
Statistical Spectral Analysis of Random Gramian Matrices

Davide Macagnano, Giuseppe Thadeu Freitas de Abreu, Centre for Wireless Communications, University of Oulu

In this paper we perform the statistical analysis on the spectrum of random $N\times N$ Gramian matrices of the form $\mathbf{G}_T^* \triangleq \mathbf{V}^\textup{T}\cdot\mathbf{G}_T\cdot\mathbf{V}$, where $\mathbf{G}$ and $\mathbf{G}_T$ are themselves Gramian matrices with subspace distance $\Delta(\mathbf{G},\mathbf{G}_T)$ and $\mathbf{V}$ is the diagonalizer of $\mathbf{G}$. In particular, we employ an extreme-value and asymptotic take on the theory of Gershgorin spectrum bounds to characterize the statistical structure of $\mathbf{G}_T^*$. The results reveal that even for relatively large $\Delta(\mathbf{G},\mathbf{G}_T)$, the matrix $\mathbf{G}_T^*$ is, with a high probability, brought to such a structure that can it be quickly diagonalized. This feature is exploited to design a statistically optimized and truncated variation of the Jacobi algorithm which is found to converge to the dominant eigenspace of $\mathbf{G}_T^*$ as fast as the deterministic \emph{optimal} sweeping strategy but without requiring its typical exhaustive search.

TP8b2-6
High-speed Nano-imaging Using Dynamic Mode AFM: A MAP Detection Approach

Naveen Kumar, Iowa State University; Govind Saraswat, Pranav Agarwal, University of Minnesota; Aditya Ramamoorthy, Iowa State University; Murti Salapaka, University of Minnesota

Nano-imaging has played a vital role in biology, chemistry and physics as it enables interrogation of material with sub-nanometer resolution. Primary means of achieving such atomic scale interrogation of matter are based on the principles of atomic force microscopy (AFM) and scanning tunneling microscopy (STM). However, current nano-imaging techniques are too slow to be useful in the high speed applications of interest such as studying the evolution of certain biological process over time that involve very small time scales. In this work, we present a high speed one-bit imaging technique using dynamic mode AFM with a high quality factor cantilever. We model the high quality factor cantilever system using a Markovian model which incorporates the inherent system memory due to the inter-symbol interference and the cantilever state. Next, we pose the imaging problem as one of finding the maximum a posteriori (MAP) symbol detector for this model. This is solved by adapting the BCJR algorithm for our channel model. Furthermore, we propose an improved MAP symbol detector that incorporates a learned prior from the previous scan line while detecting the features on the current scan line. Experimental results demonstrate that our proposed algorithm provides significantly better image resolution compared to current nano-imaging techniques at high scanning speed.

TP8b2-7

A Modified Total Variation Approach for Single Frequency Inverse Scattering

Hatim Alqadah, University of Cincinnati; Matthew Ferrara, Air Force Research Laboratory; Howard Fan, University of Cincinnati

The Linear Sampling Method (LSM) is a relatively novel method of solving the inverse acoustic or electromagnetic scattering problem. The linearity of the problem is an exact linear relationship that is satisfied by the far-field data and does not result from a linearization scheme such as the Born approximation and thus is valid under multiple scattering conditions. In this work we seek to improve performance by considering single frequency multi-static far-field data coupled with a spatial gradient constraint on the regularized image. The resulting total-variation-type optimization problem is then solved by means of an alternating minimization scheme.

TP8b2-8

A New Method for Moving-Average Parameter Estimation

Petre Stoica, Uppsala University; Lin Du, Jian Li, University of Florida; Tryphon Georgiou, University of Minnesota

We introduce an apparently original method for moving-average parameter estimation, based on covariance fitting and convex optimization. The proposed method is shown by means of numerical simulation to provide much more accurate parameter estimates, in difficult scenarios, than a related existing method does. We derive the new method via an analogy with a covariance fitting interpretation of the Capon beamforming from array processing. In the process, we also point out some new facts on Capon beamforming.

TP8b2-9

Clutter Covariance Matrices for GMTI MIMO Radar

Joshua Kantor, Dan Bliss, MIT Lincoln Laboratory

We examine the clutter covariance matrices for GMTI MIMO radar systems, and, in particular, discuss a potential suboptimal increase in their rank. This increase in rank will generically degrade the minimum detectable velocity (MDV) performance of a MIMO system. We first give a general theoretical analysis and then focus on simulated and experimental data for MIMO systems employing random and DDMA waveforms. We will show that for the DDMA waveforms the clutter covariance matrix in a given Doppler bin is effectively rank 1 for both the simulated and experimental data while for generic waveforms the rank is greater than 1.

TP8b2-10

Asymptotic Efficiency of Distributed Estimation from Constant Modulus Sensor Transmissions

Cihan Tepedelenlioglu, Mahesh Banavar, Andreas Spanias, Arizona State University

A distributed estimation problem using sensor networks is considered. A parameter is observed in noise by sensors, phase-modulated using complex exponentials and transmitted over a Gaussian multiple-access channel to a fusion center. The received signal is used to estimate the parameter under observation. The system is evaluated using the asymptotic variance of the estimate. The relationship between the Fisher information and the characteristic function is presented and used to show that the estimator is efficient if and only if the sensing noise is Gaussian.

TP8b2-11

Superfast Algorithm for Minimum Variance (Capon) Spectral Estimation

Larry Marple, Georgia Tech Research Institute; Majid Adeli, Huaping Liu, Oregon State University

The minimum variance spectral estimator (sometimes referenced as the Capon spectral estimator, or the minimum variance distortion-less response estimator) is a high resolution spectral estimator used extensively in practice as an alternative to the classical squared-magnitude Fourier estimator. In its original form without special algorithms, it requires order p^3 computations, in which is the minimum variance filter size. The current implementation used in practice employs a fast algorithm developed by Musicus, requiring order p^2 computations. This paper shows discoveries of additional exploitable structure to bring the computational order down to p*log_2*p, creating a superfast algorithm.

TP8b2-12

Equidistributed Sampling Sequences for Spectral Analysis

Mustafa Al-Ani, Andrzej Tarczynski, University of Westminster

In this paper we propose a sampling scheme for alias-free spectral analysis of signals. Unlike classical similar approaches, the scheme uses deterministic rather than random sampling. The properties of the generated sampling sequences resemble those of the random sampling ones in the sense of suppressing/reducing aliasing. Nonetheless, the endorsed scheme is shown to be more efficient and is better suited to practical implementation. It is demonstrated that for signals with bounded variations, the error in the obtained spectrum from N number of samples is bounded and can decay at the rate of log(N)/N.

TP8b2-13

An Online Method for Time-varying Spatial Spectrum Estimation Using a Towed Acoustic Array

Jeffrey Rogers, Jeffrey Krolik, Duke University

This paper addresses the problem of time-varying field directionality mapping (FDM) or spatial spectrum estimation in dynamic environments with a maneuverable towed acoustic array. Array processing algorithms for towed arrays are typically designed assuming the array is straight, and are thus degraded during tow ship maneuvers. In this paper, maneuvering the array is treated as a feature allowing for left and right disambiguation as well as improved resolution towards endfire. A new method for online spatial spectrum estimation is presented. The maximum likelihood of the time-varying field is solved for using a single expectation maximization step after each received data snapshot. A multi-source simulation is used to illustrate the proposed algorithm's ability to suppress ambiguous towed-array backlobes and resolve closely spaced interferers near endfire which pose challenges for conventional beamforming approaches, especially during array maneuvers.

TP8b2-14

Sample Covariance Based Estimation of Capon Algorithm Error Probabilities

Christ Richmond, MIT Lincoln Laboratory; Ramis Movassagh, Alan Edelman, Massachusetts Institute of Technology; Robert Geddes, MIT Lincoln Laboratory

The method of interval estimation (MIE) provides a strategy for mean squared error (MSE) prediction of algorithm performance at low signal-to-noise ratios (SNR) below estimation threshold where asymptotic predictions fail. MIE interval error probabilities for the Capon algorithm are known and depend on the true data covariance and assumed signal array response. Herein estimation of these error probabilities is considered to improve representative measurement errors for parameter estimates obtained in low SNR scenarios, as this may improve overall target tracking performance. A statistical analysis of Capon error probability estimation based on the data sample covariance matrix is explored herein.

TP8b2-15

An Optimal Spatio-Temporal Filter for Extraction and Enhancement of Multi-Channel Periodic Signals

Jesper Rindom Jensen, Mads Græsbøll Christensen, Søren Holdt Jensen, Aalborg University

This paper considers the problem of extracting periodic signals from multi-channel recordings containing noise and possibly other interfering periodic signals. We propose a novel spatio-temporal filtering method for the extraction of such signals. The proposed filter design can be seen as a generalization of the APES filter. While the filter is suited for extraction, it can also be used for joint direction-of-arrival and fundamental frequency estimation, and model order selection. The simulations illustrates the good performance of the proposed filter compared to a competing filter design and they show that the proposed filter is applicable to a real-life signal.

TP8b2-16

A Closed Form for False Location Injection under Time Difference of Arrival

Lauren Huie, Air Force Research Laboratory; Mark Fowler, State University of New York at Binghamton

We consider a sensor network tasked with emitter location via time-difference-of-arrival (TDOA), but which contains a rogue sensor trying to degrade location accuracy. We consider the rogue's task of maximally degrading estimation accuracy via the injection of a false report of its position. Our closed form solution gives a set of false positions that minimize the network's Fisher Information. We show how to select from this set the one false location that not only minimizes the Fisher Information but also optimizes the ability of the rogue to avoid detection and rejection of the false location by the locating network.

Track 1 – A. Communications Systems

Session: WAa1 – Cooperative Communications

Chair: ***Xiaoli Ma,*** *Georgia Institute of Technology*

WA1a-1 8:15 AM

Anti-Jamming in Broadcast Networks with Collaborative Frequency Hopping

Chengzhi Li, Huaiyu Dai, North Carolina State University; Liang Xiao, Xiamen Univeristy; Peng Ning, North Carolina State Univeristy

Wireless networks are vulnerable to jamming attacks in nature and traditional cryptography with requirement of pre-shared keys may be invalidated to address this issue. Uncoordinated Frequency Hopping (UFH) is a viable anti-jamming solution, however suffers from low communication efficiency.In this paper we propose a collaborative UFH scheme in a broadcast network, where a source node disseminates messages to all the other nodes in the network. The intuition behind our approach is to allow some nodes to serve as relays to the remaining nodes during the process so that the message propagation is significantly expedited. The number of relays has to be carefully controlled: too few relays offer litter help, while too many relays cause serious collisions, which even hinders the process. The optimal number of relays is derived to maximize the successful packet reception probability. In addition, to further reduce collisions and interference a novel method based on direct sequence spread spectrum is proposed and embedded in the UFH scheme. The performance of our approach is evaluated through both analysis and simulations, which demonstrate that our scheme admits strong jamming resistance as well as high communication efficiency.

WA1a-2 8:40 AM

Hybrid Relay Selection in Heterogenous Relay Networks

Mohamed Abouelseoud, Aria Nosratinia, University of Texas at Dallas

In this paper heterogenous networks are envisioned where multiple relays with different relaying capabilities co-exist in the network. Consider a representative case with one source, one destination, and multiple relays, where at each point in time a subset (e.g. two) of the relays can be active simultaneously. The question naturally arises: how can we select relays for service from among the collection of available relays, and what is the performance of each selection scheme. We propose and analyze several selection schemes in a two-relay scenario involving a number of ways to synchronously combine the relays.

WA1a-3 9:05 AM

Achieving Joint Diversity in MIMO Relay Networks with Low-Complexity Equalizers

Giwan Choi, Georgia Institute of Technology; Wei Zhang, Qualcomm Inc.; Xiaoli Ma, Georgia Institute of Technology

Wireless MIMO relay networks are receiving significant interest in recent years. However, most of MIMO relay techniques require high complexity equalizers and error free forwarding at the relay to collect diversity. In this paper, we design high-rate and low-complexity MIMO relay strategies that achieve joint cooperative and receive diversity by adopting power scaling strategy at the relay and linear equalizers at both the relay and the destination. Also, channel-controlled ARQ technique is combined to enable the spatial diversity. Through mathematical analysis, we show that the proposed design achieves joint diversity. The theoretical analysis is corroborated with numerical simulations.

WA1a-4 9:30 AM

Performance Analysis of MIMO Beamforming AF Relay Networks using Multiple Relay Antennas

Hyunjun Kim, Cihan Tepedelenlioglu, Arizona State University

Multiple-input multiple-output (MIMO) Beamforming (BF) performance is analyzed for amplify-and-forward (AF) fixed two-hop relay networks in independent and identically distributed (i.i.d.) Rayleigh fading channels. A novel performance bound using a new symbol-error-rate (SER) approximation is presented for AF relay networks equipped with multiple antennas at the source, relay, and destination when the channel state information (CSI) of the relay link is known at the source and destination. High signal-to-noise ratio (SNR) performance bound is also analyzed to simplify the performance via diversity and array gain expressions. Finally, the novel performance bound is compared with the optimized BF performance to show the tightness of the bound.

Session: WAa2 – Interference Management I

Chair: ***Eduard Jorswieck,*** *Technische Universität Dresden*

WA2a-1 8:15 AM

Randomized On-Off Signaling for Asynchronous Interference Channels

Kamyar Moshksar, Amir Khandani, University of Waterloo

This paper addresses a decentralized Gaussian interference channel consisting of two asynchronous users. The channel from each transmitter to each receiver is modeled by quasi-static and non-frequency selective Rayleigh fading. Users are asynchronous, meaning there exists a mutual delay between their transmitted codes. Due to the randomness of delay, no user is aware of the location of the interference burst along its code at the transmitter side. At the receiver sides, users are able to detect the presence of interference, however, due to the fact that users are not aware of each other's codebooks, no interference cancellation is performed, i.e., users treat each other as noise. From the transmitters' point of view, interference has a mixed PDF as a result of ambiguity on the start of the interference bursts. A stationary model for interference is considered by assuming the starting point of an interferer's codeword is uniformly distributed along the codeword of the other user. Due to non-ergodicity of the model, outage probability is an appropriate tool to study the network performance. All users follow a locally Randomized On-Off signaling scheme where each transmitter quits transmitting its Gaussian signal independently from transmission to transmission with a certain probability. An upper bound on the probability of outage per user is developed using entropy power inequality and a key upper bound on the differential entropy of a mixed Gaussian random variable. It is shown that by adopting the proposed scheme, the probability of outage is strictly lower than that of a scenario where both users keep transmitting their Gaussian signals known as continuous transmission.

WA2a-2 8:40 AM

Learning Based Mechanisms for Interference Mitigation in Self-Organized Femtocell Networks

Mohsin Nazir, Mehdi Bennis, Kaveh Ghaboosi, Centre for Wireless Communications, University of Oulu; Allen B. MacKenzie, Virginia Polytechnic Institute and State University; Matti Latva-aho, Centre for Wireless Communications, University of Oulu

We introduce two mechanisms for interference mit- igation, inspired by evolutionary game theory and machine learning to support the coexistence of a macrocell network and underlaid, self-organized femtocells. In the first approach, stand- alone femtocells choose their strategies, observe the behavior of other players, and make the best decision based on their instantaneous payoff, as well as the average payoff of all other femtocells. We formulate the interactions among selfish femtocells using evolutionary games and demonstrate how the system converges to an equilibrium. In the Reinforcement-Learning (RL) approach, however, information exchange among femtocells is not possible and each femtocell adapts its strategy and gradually learns by interacting with its surrounding environment through trial and error. Our investigations reveal that through learning, femtocells are able to self-organize and rely only on local information, while mitigating the interference towards the macrocell network. Moreover, a trade-off exists where faster convergence is obtained in the evolutionary case as compared to the RL approach, at the expense of more side information at the femtocells.

WA2a-3 9:05 AM

Spectrum Allocation and Power Control in OFDM-Based Cognitive Radios with Target SINR Constraints

Dimitrie C. Popescu, Deepak R. Joshi, Old Dominion University; Octavia A. Dobre, Memorial University of Newfoundland

In this paper we consider OFDM-based cognitive radios (CR) with linear precoders at the transmitter and minimum mean square error (MMSE) filters at the receiver, and formulate joint spectral allocation and power control subject to specified target signal-to-interference+noise ratio (SINR) as a constrained optimization problem. We discuss necessary and sufficient conditions for the optimal solution of this problem and present an algorithm that incrementally adapts the OFDM transmitter to reach the optimal transmit precoder and power values where the specified target SINR is achieved with minimum power.

WA2a-4 **9:30 AM**

Weighted Sum-Rate Maximization for a Set of Interfering Links via Branch and Bound

Chathuranga Weeraddana, Marian Codreanu, Matti Latva-aho, University of Oulu; Anthony Ephremides, University of Maryland

We consider the problem of weighted sum-rate maximization (WSRMax) for an arbitrary set of interfering links. This problem is known to be NP-hard; therefore, it is extremely difficult to solve even for a relative small number of links. The main contribution of this paper is to provide a solution method, based on the branch and bound technique, which solves WSRMax problem with an optimality certificate. At each step of the proposed algorithm, an upper and a lower bound are computed for the optimal value of the underlying problem. The algorithm terminates when the difference between the upper and the lower bound is smaller than a specified tolerance. The proposed method allows to evaluate the performance loss encountered by all heuristic methods which have been previously proposed to obtain suboptimal solutions for the same problem. The proposed algorithm is also useful to provide performance benchmarks by back-substituting it into many existing network design problems which relies on solving WSRMax problem.

Track 3 – C. Networks

Session: WAa3 – Sensor Networks

Chair: ***Milica Stojanovic,*** *Massachusetts Institute of Technology*

WA3a-1 **8:15 AM**

Sensor Scheduling for Energy-Efficient Target Tracking in Sensor Networks

George Atia, Jason Fuemmeler, Venugopal Veeravalli, University of Illinois at Urbana-Champaign

In this paper we study the problem of tracking an object moving randomly through a network of wireless sensors. Our objective is to devise strategies for scheduling the sensors to optimize the tradeoff between tracking performance and energy consumption. We cast the scheduling problem as a Partially Observable Markov Decision Process (POMDP) where the control actions correspond to the set of sensors to activate at each time step. Using a bottom-up approach, we consider different sensing, motion and cost models with increasing level of difficulty. At the first level, the sensing regions of the different sensors do not overlap and the target is only observed within the sensing range of an active sensor. Then, we consider sensors with overlapping sensing range such that the tracking error, and hence actions for different sensors, are tightly coupled. Finally, we consider scenarios wherein target locations and sensors' observations assume values on continuous spaces. An exact solution is generally intractable even for the simplest model due to the dimensionality of the information and action spaces. Hence, we devise approximate solution techniques and in some cases derive lower bounds on the optimal tradeoff curves. Even though suboptimal, the generated scheduling policies often provide close-to-optimal energy-tracking tradeoffs.

WA3a-2 **8:40 AM**

Clustered Ad-Hoc Networks in the Presence of Interference

Andrej Stefanov, Milica Stojanovic, Northeastern University

We consider the performance of clustered underwater acoustic ad-hoc networks in the presence of interference. We assume a uniform distribution of nodes over a finite area. The cluster-to-cluster channel is modeled through the Ricean fading model. We adopt a communication theoretic approach and study the interdependence of the sustainable number of cluster-to-cluster hops through the network, end-to-end frame error probability, power and bandwidth allocation. We find that the network's ability to provide full connectivity may be limited from below by coverage and from above by interference. In other words, the network may be coverage-limited, when the number of nodes in the network is small, and interference-limited, when the number of nodes is high. We present numerical examples that illustrate the results of the analysis.

WA3a-3 **9:05 AM**

Maximizing Lifetime in Wireless Sensor Networks Under Opportunistic Routing

Michal Kaliszan, Slawomir Stanczak, Fraunhofer German-Sino Lab for Mobile Communications

The main design challenge in wireless sensor networks is to achieve satisfactory network lifetimes under scarce energy resources available at the nodes. In this paper we present an optimization framework for maximizing lifetime of a network in which opportunistic routing together with random linear network coding is used. We propose a scheme in which each node attempts to receive transmissions from a given neighbor, and thus consumes energy for receiving, with a certain probability. Optimality conditions are presented and approaches to solving the problem are discussed. We conclude with numerical experiments which confirm performance improvements achieved using our framework.

WA3a-4 **9:30 AM**

Distributed Averaging in Wireless Sensor Networks Under an ALOHA-like Communication Protocol

Valentin Schwarz, Gerald Matz, Vienna University of Technology

Distributed averaging is a fundamental tool for inference in wireless sensor networks; it builds on the exchange of messages between nodes to achieve consensus. Practical implementations of distributed averaging require a communication protocol that manages the message exchange. We propose a simple asynchronous ALOHA-like protocol that requires no node coordination and applies to any averaging scheme that tolerates link failures. We analytically characterize the collisions occurring with this protocol and we provide numerical performance results for asynchronous distributed averaging under the proposed protocol.

Track 4 – D. Adaptive Systems and Processing

Session: WA4 – Advances on Adaptive Filtering and Applications

Co-Chairs: ***Jerónimo Arenas-García,*** *Universidad Carlos III de Madrid* ***and Magno Teofilo Madeira da Silva,*** *University of Sao Paulo*

WA4-1 **8:15 AM**

Nonlinear Adaptive Filtering via Soft Clustered Linear Models

Andrew C. Singer, Kyeongyeon Kim, Jun Won Choi, University of Illinois; Suleyman Serdar Kozat, Koc University

Estimation methods for nonlinear systems based on locally linear models have been widely used due to their mathematical and algorithmic simplicity. In competitive neural networks, for example, the simplest such locally linear algorithm, Winner-Take-All (WTA), updates only one locally linear model at a time during training. After training, the output estimate may be selected from one of the local models or be given by a combination of the K-best local model outputs. For a nonlinear dyanmic system, output samples that are close in an appropriate sense in the dynamic feature domain are often far apart in the time series. As such, temporal updates may occur too infrequently to track temporal fluctuations in the dyamics. In this paper, we model a nonlinear system using N linear adaptive filters based on a soft clustering of features from the observations. To tradeoff both slow and fast varying features, we update K association probabilities among N clusters during clustering. Based on this soft clustering, we update K linear filters per data sample and combine the outputs to minimize squared error. In this paper, the proposed algorithm is applied to adaptive turbo equalization (TEQ), where the soft clustering based approach is compared with conventional adaptive TEQ and adaptive TEQ using conventional local models.

WA4-2 **8:40 AM**

Sparsity-Cognizant Subspace Tracking for Dimensionality Reduction

Ioannis Schizas, Georgios B. Giannakis, University of Minnesota

Subspace tracking emerges in a number of signal processing tasks including data compression, direction of arrival, and blind channel estimation. Sparsity is an attribute characterizing many natural and man-made signals. Sparse structures may also be present in signal subspaces, or covariance eigenspaces whose basis vectors contain a large portion of zero entries. For instance, the covariance eigenspace of random spatial fields could exhibit a sparse structure due to the uncorrelatedness of field measurements acquired across different spatial locations. The sparsity pattern of the subspace basis vectors may also be time-varying due to the nonstationarity of the sources creating the field. In this context, a novel online data processing algorithm is developed to track the support of subspace basis vectors, and also estimate accurately their nonzero entries. Relying on a l1-norm regularized exponentially weighted least-squares cost and employing coordinate descent, computationally efficient updating recursions are derived for tracking the sparse subspace basis vectors. The sparse subspace tracking scheme is tested in compressing nonstationary signals. Numerical examples demonstrate that the novel approach outperforms existing sparsity-agnostic alternatives such as the projection approximation subspace tracking algorithm.

WA4-3 **9:05 AM**

Bacterial Motility via Diffusion Adaptation

Jianshu Chen, Xiaochuan Zhao, Ali H. Sayed, University of California, Los Angeles

Bacteria forage by moving towards nutrient sources or chemical stimuli in a process known as Chemotaxis. The bacteria follow gradient variations in one of two modes: they either move in straight lines or tumble. Both modes of movement are affected by Brownian motion. In this paper, we further assume that the bacteria are capable of limited interaction through the emission of chemical signals. As the bacteria swim towards the highest concentration of nutrients, they emit chemical signals that are sensed by the neighboring bacteria. The aggregate effect of the chemicals in a spatial region can be used by the bacteria to adjust their direction of motion. In this way, the bacteria are able to respond to local stimuli based on the nutritional gradient and to neighborhood information based on the chemical signals emitted by the neighboring bacteria. In this paper, we study different schemes for cooperation and diffusion of information and examine their effect on bacteria motility. Because bacteria are limited

in their abilities, we restrict the sharing of information to binary choices (such as whether to run or tumble). We also examine the accumulation and decay effect of the chemical signals. Simulation results suggest that cooperation among bacteria is critical for effective foraging to improve their decisions of movement. The diffusion algorithms used in this study belong to the class of adaptive diffusion methods for distributed estimation, with the added constraint of limited information sharing.

WA4-4 9:30 AM

Adaptive Reduced-Rank Least Squares Beamforming Algorithm Based on The Set-Membership Framework

Lei Wang, Rodrigo C. de Lamare, The University of York

This paper presents a new adaptive algorithm for the linearly constrained minimum variance (LCMV) beamformer design. We incorporate the set-membership filtering (SMF) framework into the reduced-rank joint iterative optimization (JIO) scheme to develop a constrained recursive least squares (RLS) based algorithm called JIO-SM-RLS. The proposed algorithm inherits the positive features of reduced-rank signal processing techniques to enhance the performance, and utilizes the data-selective updates (around 10-15%) of the SMF methodology to save the computational cost significantly. An effective time-varying bound is imposed on the array output as a constraint to circumvent the risk of overbounding or underbounding, and to update the parameters for beamforming. The updated parameters construct a set of solutions (a membership set) that satisfy the constraints of the LCMV beamformer. Simulations are performed to show the superior performance of the proposed algorithm in terms of convergence rate and reduced computational complexity over the existing methods.

BREAK 9:55 AM

WA4-5 10:15 AM

Advances in Identification and Compensation of Nonlinear Systems by Adaptive Volterra Models

Marcus Zeller, Walter Kellermann, University of Erlangen-Nuremberg

Today, there exist a number of applications where the usual assumption of technical systems with purely linear input/output relation is violated by the presence of nonlinear distortions. For instance, the contradicting demands of modern communications systems, namely wide dynamic ranges along with efficient use of transmission power, and the ongoing pressure of miniaturization of electro-acoustic/electro-mechanic hardware components leads almost inevitably to nonnegligible saturation effects. It is widely acknowledged that the Volterra filter structure is a very general model for describing a large class of nonlinear distortions with either memoryless or dispersive characteristics. Although many authors have also focussed a considerable amount of research towards less demanding alternative nonlinear models, these approaches are often based on application-specific constraints and cannot easily be translated to arbitrary problems. In this contribution, we present the recent advances in modeling and compensation of nonlinearities by adaptive Volterra systems. In particular, we address the possibility of realizing the filtering more efficiently in the DFT domain by resorting to a MISO (multiple-input single-output) representation. In order to accelerate the speed of convergence, it has been shown that it is useful to exploit the available signal energy more often in an iterated filtering/updating manner. Moreover, methods for further reducing the required number of operations for time-domain implementations are discussed. As the most promising scheme, we present a technique that can be employed so as to reliably estimate and control the memory of the quadratic kernel of a second-order nonlinear model, thus realizing an adaptive filter structure with dynamically growing/shrinking size. Since this generic evolutionary principle can be extended to linear and higher-order filters as well, future work will aim at implementing a generic class of adaptive filters that can be used for arbitrary (i.e. linear and/or nonlinear) system identification tasks. Although these schemes hold for any use case of Volterra filters, we show the effectiveness of the proposed algorithms by considering a highly challenging nonlinear acoustic echo cancellation scenario, calling for system identification of time-varying kernels with large sizes excited by nonstationary and nonwhite (speech) signals.

WA4-6 10:40 AM

Adaptive Pre-distortion Techniques Based on Orthogonal Polynomials

Robert Dallinger, Vienna University of Technology; Henri Ruotsalainen, Risto Wichman, Aalto University School of Science and Technology; Markus Rupp, Vienna University of Technology

Pre-distortion in digital baseband is a cost-effective method to linearise the power amplifiers used in the transmitters of spectrally efficient communication systems. From a signal processing point of view, the decision which model to choose as well as the selected adaptive algorithm primarily determine the functional structure of the pre-distorter. During the last two decades, in literature, a multitude of pre-distorter structures has been proposed and analysed. In this work, we focus on simple and commonly employed models (the simplified Wiener model and the Hammerstein model) consisting of a linear filter and a static nonlinearity. The latter is represented using a basis of orthogonal polynomials. The models are adapted by well known direct respectively indirect learning structures. Based on burst measurements with a commercial power amplifier and practical transmission signals, this work compares the real-world performance of such systems and investigates how they are influenced by the choice of the orthogonal polynomial basis.

WA4-7 11:05 AM

PtNLMS Algorithm Obtained by Minimization of Mean Square Error Modeled by Exponential Functions

Kevin Wagner, Naval Research Laboratory; Miloš Doroslovački, The George Washington University

Using the proportionate-type steepest descent algorithm we represent the current weight deviations in terms of initial weight deviations. Then we attempt to minimize the mean square output error with respect to the gains. The corresponding optimal gains are obtained using a water-filling procedure. The stochastic counterpart is obtained by replacing the true weights, which are unknown, by their current estimates. Additionally, a simplified gain allocation method is proposed. The resulting algorithm behaves initially like the proportionate normalized least mean square algorithm and as time proceeds the algorithm behaves like the normalized least mean square algorithm. This type of behavior is typically desired and results in enhanced convergence performance. We present results for the new algorithm and compare it to other standard proportionate type normalized least mean square algorithms.

WA4-8 11:30 AM

Iterative State Estimation

Thomas J. Riedl, Andrew C. Singer, University of Illinois at Urbana-Champaign

This paper presents an iterative method for the estimation of the internal states of a given discrete-time linear state-space model from a series of noisy measurements. In particular we identify the MAP estimate of those states as being the solution of a sparse system of linear equations and derive an iterative solver based on the conjugate gradient method. We derive convergence results that allow for a trade-off between speed and accuracy and finally apply the method to channel estimation where it is shown to outperform kalman smoothing complexity-wise.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: WA5 – Statistical Signal Processing

Chair: ***Daniel Fuhrmann,*** *Michigan Technological University*

WA5-1 8:15 AM

Biologically Inspired Coupled Antenna Array for Direction of Arrival Estimation

Murat Akcakaya, Washington University in St. Louis; Carlos H. Muravchik, Universidad Nacional de La Plata; Arye Nehorai, Washington University in St. Louis

Inspired by the female Ormia ochracea's mechanically coupled ears, we propose to design a small-size transmission coupled antenna array with high performance radiation pattern. The mechanical coupling provides the female Ormia with high localization accuracy despite the small distance between its ears compared with the incoming wavelength of the source signal. The mechanical coupling between the Ormia's ears has been modeled by a pair of differential equations. In this paper, after solving these differential equations governing the Ormia ochracea's ear response, we convert the response to pre-specified radio frequencies. Using the converted response, we then implement the biologically inspired coupling as a multi-input multi-output filter on a uniform linear antenna array output. We derive the maximum likelihood estimates of source direction of arrivals (DOAs), and compute the corresponding Cram\'{e}r-Rao bound on the DOA estimation error as a performance measure. We use Monte Carlo numerical examples to demonstrate the advantages of the coupling effect.

WA5-2 8:40 AM

Exploiting a Constellation of Narrowband RF Sensors to Detect and Track Moving Targets

Chris Kreucher, Integrity Applications Incorporated; J. Webster Stayman, Johns Hopkins University; Ben Shapo, Integrity Applications Incorporated; Mark Stuff, Michigan Tech Research Institute

This paper presents a novel approach to detecting and tracking moving targets using a constellation of narrowband radio frequency sensors. Our methodology is an innovative combination of nonlinear estimation and information theoretic sensor placement. The nonlinear filtering approach fully exploits bistatic Doppler measurements made by the sensors without thresholding or linear/Gaussian assumptions, thereby improving the detection/false alarm tradeoff and lowering tracking error. The information theoretic sensor placement algorithm, which is based on minimizing the Cramer-Rao bound on localization variance, selects sensor positions that lead to the best estimation performance, thereby maximally exploiting the finite sensing resources.

WA5-3 **9:05 AM**

On the Use of Mismatched Wiener Filtering for the Characterization of Non-stationary Channels

Adrian Ispas, RWTH Aachen University; Laura Bernadó, Telecommunications Research Center Vienna; Meik Dörpinghaus, Gerd Ascheid, RWTH Aachen University; Thomas Zemen, Telecommunications Research Center Vienna

We develop a method for the determination of local regions in time in which a channel can be approximated as stationary. Contrary to previous results in literature relying on to some extent arbitrary measures and thresholds, we consider a realistic (flat fading) channel estimator and relate the size of local quasi-stationarity regions to the degradation of the mean squared channel estimation error. Furthermore, we obtain an approximate, but much simplified, evaluation of the mean squared error. We exemplarily evaluate both measures using realistic channel measurements, and find that their results show the same trends.

WA5-4 **9:30 AM**

A Lower Bound on the Estimator Variance for the Sparse Linear Model

Sebastian Schmutzhard, University of Vienna; Alexander Jung, Franz Hlawatsch, Vienna University of Technology; Zvika Ben-Haim, Yonina C. Eldar, Technion - Israel Institute of Technology

We study the performance of estimators of a sparse nonrandom vector based on an observation which is linearly transformed and corrupted by additive white Gaussian noise. Using the reproducing kernel Hilbert space framework, we derive a new lower bound on the estimator variance for a given differentiable bias function (including the unbiased case) and an almost arbitrary transformation matrix (including the underdetermined case considered in compressed sensing theory). For the special case of a sparse vector corrupted by white Gaussian noise—i.e., without a linear transformation—and unbiased estimation, our lower bound improves on previously proposed bounds.

BREAK **9:55 AM**

WA5-5 **10:15 AM**

Knowledge-aided Parametric GLRT for Space-Time Adaptive Processing

Pu Wang, Hongbin Li, Stevens Institute of Technology; Braham Himed, Air Force Research Laboratory

In this paper, we consider knowledge-aided space-time adaptive processing (KA-STAP) with a parametric approach, where disturbances in both test and training signals are modeled as a multichannel auto-regressive (AR) model. The a priori knowledge is incorporated into the detection problem through a stochastic signal model, where the spatial covariance matrix of the disturbance is assumed random. According to this model, a Bayesian version of the parametric generalized likelihood ratio test (PGLRT) is developed in a two-step approach, which is referred to as the KA-PGLRT. Interestingly, the KA-PGLRT employs a colored loading approach for estimation of the spatial covariance matrix of the test signal. Simulation results show that the KA-PGLRT can obtain better detection performance over other parametric detectors.

WA5-6 **10:40 AM**

Joint Estimation of Target Reflectivity and Local Oscillator Phases in a MIMO Radar Systems with Distributed Assets

Changyu Sun, Daniel Fuhrmann, Michigan Technological University

Synchronization across multiple distributed apertures is the standard assumption in proposed multi-input multi-output (MIMO) radar systems. However, perfect phase synchronization is difficult to realize. Assuming frequency synchronization, probably through a GPS system or reception of a beacon, we address the issue of unknown phase mismatches of local oscillators in a MIMO radar system, and consider the problem of joing estimation of target reflectivity and phase mismatches. The bistatic target reflectivity is subject to a complex Gaussian prior distribution given by a multiple point scatterer model and phase mismatches are assumed independent and identically distributed random variables.

WA5-7 **11:05 AM**

Comparison of Nonparametric and Parametric Time-Varying Methods for Quantifying Phase Synchrony

Ali Mutlu, Selin Aviyente, Michigan State University

Quantifying the pairwise relationships between two signals is an important problem in many fields of science and engineering. Phase synchrony has been suggested as a powerful tool for determining bivariate signal properties. Recently, a nonparametric phase estimation method based on the Rihaczek distribution belonging to Cohen's class of time-frequency distributions (TFDs)

has been proposed for the estimation of time-varying phase. Alternatively, parametric time-frequency methods, which model the phase as a polynomial, can be used. In this paper, the nonparametric phase estimator based on the Rihaczek distribution is compared with the parametric estimator based on the high-order ambiguity function for their accuracy in estimating phase synchrony.

WA5-8 11:30 AM

Maximum-Likelihood and Best Invariant Orientation Estimation

Ian Clarkson, University of Queensland; Stephen Howard, Defence Science & Technology Organisation; William Moran, University of Melbourne; Douglas Cochran, Arizona State University; Megan Dawson, University of Queensland

Estimation of orientation or attitude is a longstanding problem in statistics and signal processing, with applications to crystallography, aeronautics and computer vision. In this paper, we consider the Mackenzie-Wahba estimator. This estimator is known to be maximum-likelihood in a certain sense but we further show that it is a Bayes estimator and also a best invariant estimator.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: WAa6 – Estimation and Detection

Chair: ***Jun Zhang,*** *Arizona State University*

WA6a-1 8:15 AM

Joint Map Estimation and Localization using Distance Measurements to Landmarks with Unknown Location

Andreas Richter, Aalto University

In this paper the problem of joint localization and map estimation is studied. The objective is to determine the location of a set of fixed nodes, e.g. access-points or base-stations in a wireless network using only distance measurements acquired by a node moving in the vicinity of the fixed nodes. It is assumed, that no prior information about locations of any of the nodes (fixed or mobile) is available, i.e. no anchor nodes exist. It is shown that the map of the fixed nodes as well as the locations of the mobile node(s) in this map can be identified unambiguously from ranging measurements. A closed form algorithm is derived to estimate the map. Furthermore, the performance of the derived estimator is compared in Monte-Carlo simulations with (i) the Cram\'er-Rao lower bound and (ii) with the performance of an iterative maximum-likelihood estimator.

WA6a-2 8:40 AM

Distributed Detection over Gaussian Multiple Access Channels with Constant Modulus Signaling

Cihan Tepedelenlioglu, Sivaraman Dasarathan, Arizona State University

A distributed detection scheme where the sensors transmit with constant modulus signals over a Gaussian multiple access channel is considered. The deflection coefficient (DC) of the proposed scheme is shown to depend on the characteristic function of the sensing noise and the error exponent for the system is derived using large deviation theory. Optimization of the DC and error exponent are considered with respect to a transmission phase parameter for a variety of sensing noise distributions including impulsive ones. The proposed scheme is also favorably compared with existing amplify-and-forward and detect-and-forward schemes. Simulations corroborate that the DC and error exponent can be effectively used to optimize the error probability for a wide variety of sensing noise distributions.

WA6a-3 9:05 AM

Matching Pursuits May Yield Superior Results to Orthogonal Matching Pursuits When Secondary Information is Estimated From the Signal Model

Guifeng (Rick) Liu, Victor DeBrunner, Florida State University

In this paper, we compare the Orthogonal Matching Pursuits [1] and the Matching Pursuits [2] algorithm as used with the same dictionary in a stationary spectral estimation problem. Specifically, we develop an over-complete dictionary using the Fourier and Hirschman Optimal [3] Transforms. Then, we apply a periodogram spectral estimation algorithm using this dictionary to a signal consisting of closely-spaced-in-frequency sinusoids in additive white noise. It is well-known that the matching pursuit algorithm is not guaranteed to converge as more dictionary elements are used to represent the signal, the orthogonal version must. What is not well-known, at least to the authors, is that the frequency estimation (i.e. the secondary inference that is desired from the signal

model) is not highly connected to the modeling performance. In fact, we will show that while the energy in the residual is lower for the orthogonal matching pursuits, the frequency estimation error is lower for the spectral estimated derived using matching pursuits.

WA6a-4 9:30 AM

Adaptive Sensing and Target Tracking of a Simple Point Target with Online Measurement Selection

Aashish Poudel, Daniel Fuhrmann, Michigan Technological University

In previous work (Asilomar 2008) we considered the problem of estimating target location parameters using an adaptive sensing paradigm, when there is an unknown complex amplitude in the target response. Here we extend that work to target tracking. At each step in a discrete-time extended Kalman filter, a measurement matrix is chosen to illuminate optimally the target location response and its tangent plane in the array response manifold, based on prior position and velocity information from the previous step. Performance improvements relative to non-adaptive target tracking methods will be quantified.

Track 8 – H. Speech, Image and Video Processing

Session: WAa7 – Sparse Representations in Image Processing

Chair: ***Shubha Kadambe,*** *Rockwell Collins*

WA7a-1 8:15 AM

Compressive Sensing and Vector Quantization Based Image Compression

Shubha Kadambe, Rockwell Collins Inc.

Intelligent Surveillance and Reconnaissance (ISR) collection systems use Unmanned Aerial Vehicles (UAVs) with different imaging sensors. They suffer from limited communication bandwidth and computation resources. To ameliorate this, we propose a novel compression technique by combining inversion scheme of Compressive Sensing (CS), Vector Quantization (VQ) and Arithmetic Coding (AC). We have applied it to compress images and videos, and compared it with JPEG/MPEG. Our results indicate that our algorithm provides better quality images at the same compression rate (or two times better compression ratio) and eleven times faster compression of images as compared to JPEG/MPEG. Complete details and comparison results will be provided in the full paper.

WA7a-2 8:40 AM

Image Sequence Change Detection via Sparse Representations

Andrew Lingg, Wright State University; Edmund Zelnio, Air Force Research Laboratory; Frederick Garber, Brian Rigling, Wright State University

We present a sparsity-based algorithm for image change detection. A series of reference images in conjunction with an orthonormal basis are used to form an overcomplete basis. The mission image can then be represented as a linear combination of reference images and components of the orthonormal basis. We solve for the linear combination that has minimum l^1 norm, encouraging a sparse representation. Components of the solution that lie within the orthonormal basis indicate changes in the mission image. We discuss experiments using this algorithm and the results that were obtained.

WA7a-3 9:05 AM

Parameterized Deformation Sparse Coding via Tree-Structured Parameter Search

Brandon Burdge, Kenneth Kreutz-Delgado, Joseph Murray, University of California, San Diego

Representing transformation invariances in data is known to be valuable in many domains. We consider a method by which prior knowledge about the structure of such invariances can be exploited using a novel algorithm for sparse coding across a learned dictionary of atoms combined with a parameterized deformation function that captures invariant structure. We demonstrate the value of this on both reconstructing signals, as well as improved unsupervised grouping based on invariant sparse representations.

WA7a-4 9:30 AM

A Unified FOCUSS Framework for Learning Sparse Dictionaries and Non-squared Error

Brandon Burdge, Kenneth Kreutz-Delgado, Joseph Murray, University of California, San Diego

FOCUSS is an Iteratively Reweighted Least Squares approximation used to find the inverse solution of an underdetermined linear system when the source vector is assumed to be sparse. It also provides an iterative descent method used to solve for an unknown dictionary. We describe three extensions to the FOCUSS model: First a choice of generalized p-norm reconstruction error which corresponds to differing assumptions on the cost of errors. Second the use of a constraint which encourages sparsity

on the dictionary atoms, and third the combination of both sparsity on dictionary atoms and generalized reconstruction error to form one unified framework for solving a wide set of sparsity requirements on sources, on loadings, and on error. Finally, we describe a practical set of algorithms for learning dictionaries and source vectors under each of these model assumptions, and show experimental results using these algorithms.

Track 1 – A. Communications Systems

Session: WAb1 – Communication Theory

Chair: ***Visa Koivunen,*** *Aalto University*

WA1b-1 10:15 AM

Equivocation of Eve Using Two Edge Type LDPC Codes for the Erasure Wiretap Channel

Vishwambhar Rathi, Mattias Andersson, Ragnar Thobaben, Royal Institute of Technology (KTH); Joerg Kliewer, New Mexico State University; Mikael Skoglund, Royal Institute of Technology (KTH)

We consider transmission over the binary erasure wiretap channel. We use two edge type LDPC codes based on the coding scheme of Thangaraj et al. By generalizing the method of Measson, Montanari, and Urbanke, we compute the equivocation of the wiretapper for two edge type LDPC codes.

WA1b-2 10:40 AM

Achievable Rates in Two-user Interference Channels with Finite Inputs and (Very) Strong Interference

Frederic Knabe, Aydin Sezgin, Ulm University

We consider the two-user interference channel in the strong and very strong interference regime. In those regimes the capacity is known. However, to achieve this capacity, the channel inputs need to be Gaussian distributed and as such are not well suited for practical applications. In this paper, we investigate the achievable rates if the channel inputs are restricted to finite constellations. We show that rotating one of these input alphabets in the complex plane can increase the achievable rate region. Furthermore, we show that the threshold at which the single-user rates are achieved also depends on this rotation.

WA1b-3 11:05 AM

On the Optimality of Channel Inversion with Diversity

Yuan Zhang, Cihan Tepedelenlioglu, Arizona State University

Asymptotic comparisons of ergodic channel capacity at high SNR are provided for optimal power and rate adaptation and channel inversion over fading channels with general distributions, as well as the capacity of the equivalent AWGN channel. We present closed-form expressions for the high-SNR capacity difference gaps among these capacities. Based on these expressions it is shown that the presence of space diversity or multi-user diversity make channel inversion arbitrarily close to achieving optimal capacity at high SNR with sufficiently large number of antennas or users. Numerical results are shown to corroborate our analytical results.

WA1b-4 11:30 AM

Outage Analysis for Hybrid Relaying in the Parallel Relay Network

Samantha Summerson, Behnaam Aazhang, Rice University

Cooperative communication improves wireless communications by providing diversity, but optimal relay selection remains a relatively open problem. We analyze this problem in a network which employs multiple parallel relays. Relays are selected to participate based on instantaneous channel magnitudes, so the set of active relays in a block of time is stochastic. We define an opportunistic hybrid relaying scheme composed of two relay protocols and derive thresholds which determine forwarding behaviors. These thresholds are based on outage probabilities. Numerical results for end-to-end outage probability are presented, demonstrating that the flexibility of our protocol offers performance gains over single-mode relaying protocol.

Session: WAb2 – Interference Management II

Chair: ***Anna Scaglione,*** *University of California at Davis*

WA2b-1 10:15 AM

A Study on the Optimal Degrees-of-Freedom of Cellular Networks: Opportunistic Interference mitigation

Bang Chul Jung, Gyeongsang National University; Dohyung Park, Samsung Electronics Co., Ltd.; Won-Yong Shin, Harvard University

An opportunistic interference mitigation (OIM) is introduced for cellular networks with time-invariant channel coefficients and multi-antenna base stations (BSs), where a user scheduling strategy is utilized in uplink communication environments. We consider two OIM protocols, where each BS opportunistically selects users who generate the minimum interference to the other BSs. Then, their performance is analyzed in terms of degrees-of-freedom (DoFs), and it is proved that our scheme is DoF-optimal. Numerical evaluation is also shown.

WA2b-2 10:40 AM

Transport Capacity for Networks of Interfering Multiple-Access Channels

Christian Peel, Pengcheng Zhan, ArrayComm, LLC

We analyze the distance-weighted rate achievable in wireless networks that utilize multiple-access links. We give a centralized algorithm for link activation, which gives results close to the optimal link activation graph. We derive an analytic upper bound on the capacity and prove that in networks with an asymptotically large number of nodes the throughput grows with the number of antennas, and depends on the expected path loss between nodes. Numerical examples agree with the analysis and show the benefit of multiple-access links over point-to-point links even in the presence of significant interference.

WA2b-3 11:05 AM

Interference Management through Mobile Relays in Ad Hoc Networks

Rohit Naini, Pierre Moulin, University of Illinois at Urbana-Champaign

Adhoc Networks nodes engage in localized grouping and organization based on their neighbourhood to carry out complex goals such as end to end communication. Certain network nodes are enlisted as localized relays to assist in passing messages along a chain. This paper proposes a method to exploit the presence of relay nodes in wireless networks to mitigate interference from other simultaneous transmissions. Optimal spatial locations and transmission schemes to combat interference are identified for cognizant mobile relays.

WA2b-4 11:30 AM

Interference Alignment Through Staggered Antenna Switching for MIMO BC With No CSIT

Chenwei Wang, Tiangao Gou, Syed Jafar, University of California, Irvine

In this paper, we explore the degrees of freedom (DoF) of the broadcast channel (BC) where the transmitter is equipped with M antennas and there are K receivers, each equipped with N reconfigurable antennas capable of switching among M preset modes. Without any knowledge of the channel coefficient values but only receiver antenna switching modes at the transmitter, we propose an interference alignment scheme for this channel. We show that if $N<M$, then a total of $MNK/(M+KN-N)$ DoF are achievable, almost surely. The key to this interference alignment scheme is the ability of the receivers to switch between reconfigurable antenna modes to create short term channel fluctuation patterns that are exploited by the transmitter. Compared to the results we showed for MISO BC [6], the supersymbol of MIMO BC may have diverse structures depending M and N, and it can be determined from that of MISO BC through an iterative mapping function.

Track 2 – B. MIMO Communications and Signal Processing

Session: WAb3 – Multiuser Beamforming and Interference Channels

Chair: ***Dan Bliss,*** *MIT Lincoln Labs*

WA3b-1 10:15 AM

A Robust and Efficient Transmission Technique for the LTE Downlink

Gerhard Wunder, Jan Schreck, Fraunhofer MCI, Heinrich-Hertz-Institut

We present a new robust and efficient feedback and transmit strategy for multiuser MIMO downlink communication systems, termed Rate Approximation, and analyze its performance. It is shown that the rate gap due to the rate--constrained feedback channel scales double exponentially with the number of feedback bits and that the optimal throughput scaling can be achieved, asymptotically. The results are sustained by system level simulations.

WA3b-2 10:40 AM

Robust Transceiver Design for K-Pairs Quasi-Static MIMO Interference Channels via Semi-Definite Relaxation

Eddy Chiu, Vincent K. N. Lau, Hong Kong University of Science and Technology; Tao Wu, Sheng Liu, Huawei Technologies, Co. Ltd.

In this paper, we propose a robust transceiver design for the K-pair quasi-static MIMO interference channel. Each transmitter is equipped with M antennas, each receiver is equipped with N antennas, and the k-th transmitter sends L_k independent data streams to the desired receiver. In the literature, there exist a variety of theoretically promising transceiver designs for the interference channel such as interference alignment-based schemes, which have feasibility and practical limitations. In order to address practical system issues and requirements, we consider a transceiver design that enforces robustness against imperfect channel state information (CSI) as well as fair performance among users.

WA3b-3 11:05 AM

MIMO Interference Channel with Confidential Messages: Game Theoretic Beamforming Designs

Ali Fakoorian, A. Lee Swindlehurst, University of California, Irvine

We study the achievable rate regions of the multiple-input multiple-output (MIMO) interference channel with confidential messages sent to two receivers. Under this model, each receiver is an eavesdropper for the other link and tries to decode both his own message and the message intended for the other receiver. We describe several transmission schemes for Gaussian interference channels and derive their achievable secrecy rate regions under the assumption that the transmitters use low complexity beamforming algorithms. We also consider the trade-offs associated with their need for channel state information at the transmitter (CSIT). We approach the above topics from a game-theoretic perspective and we consider the resulting operating points from the viewpoint of fairness and efficiency

WA3b-4 11:30 AM

On Duality in the MISO Interference Channel

Francesco Negro, Eurecom; Irfan Ghauri, Infineon France; Dirk Slock, Eurecom

SINR duality is shown in a multi-input single-output (MISO) interference channel (IFC) and its dual SIMO with linear transmit (Tx) beamformers (BF). While uplink (UL) downlink (DL) duality for the SINR balancing (max min SINR) beamforming problem under the sum power constraint is well-established between the Broadcast channel (BC) and its (easier to solve) UL Multiple Access (MAC) dual channel, such duality does not at first seem relevant for the IFC. We show that SINR duality under the sum power constraint nevertheless holds in the MISO IFC leading to BF design through similar considerations as the BC-MAC case. We next impose further per-Tx power constraints meaningful for the IFC structure and show continued existence of SINR duality in the MISO IFC and the corresponding UL SIMO dual channel, but this time with a different UL noise. The beamformers, Tx powers and noise variances are found through an iterative algorithm.

Session: WAb6 – SOC Architectures and Applications

Chair: ***E. Deprettere,*** *Leiden University*

WA6b-1 10:15 AM

PREcision Timed (PRET) Machine

Isaac Liu, Edward A. Lee, University of California, Berkeley

This paper proposes an paradigm shift in computer architecture design for real-time cyber-physical systems. Conventional techniques employed in computer architectures give average case performance improvement at the cost of unpredictable timing behavior, which leads to difficulties in analysis and design of the timing properties and composition of systems. We introduce PRET (Precision Timed Machine), a computer architecture designed with a focus on timing predictability and composability. PRET employs thread-interleaved pipelines and scratchpads to gain timing predictability without sacrificing performance. We show that PRET improves the precision of timing analysis methods for real-time cyber physical systems.

WA6b-2 10:40 AM

Time-predictable Chip-Multiprocessor Design

Martin Schoebrl, Technical University of Denmark

Real-time systems need time-predictable platforms to enable static worst-case execution time (WCET) analysis. Improving the processor performance with superscalar techniques makes static WCET analysis practically impossible. However, most real-time systems are multi-threaded applications and performance can be improved by using several processor cores on a single chip. In this paper we present a time-predictable chip-multiprocessor system that aims to improve system performance and is still WCET analyzable.

WA6b-3 11:05 AM

Design and Implementation of Real-time Signal Processing Applications on Heterogeneous Multiprocessor Arrays

Hsiang-Huang Wu, Chung-Ching Shen, University of Maryland; Michael Schulte, AMD Research and Advanced Development Labs; Tong Zhang, Rensselaer Polytechnical Institute; Shuvra Bhattacharyya, University of Maryland

Processing structures based on arrays of computational elements form an important class of architectures, which includes field programmable gate arrays (FPGAs), systolic arrays, and various forms of multicore processors. A wide variety of design methods and tools have been targeted to regular processing arrays involving homogeneous processing elements. In this paper, we introduce the concept of field programmable X arrays (FPXAs) as an abstract model for design and implementation of heterogeneous multiprocessor arrays for signal processing systems. FPXAs are abstract structures that can be targeted for implementation on applicationspecific integrated circuits, FPGAs, or other kinds of reconfigurable processors. FPXAs can also be mapped onto multicore processors for flexible emulation. We discuss the use of dataflow models as an integrated application representation and intermediate representation for efficient specification and mapping of signal processing systems on FPXAs. We demonstrate our proposed models and techniques with a case study involving the embedding of an application-specific FPXA system on an off-the-shelf FPGA device.

WA6b-4 11:30 AM

Buildings as Cyber-physical Energy Systems

Yuvraj Agarwal, Thomas Weng, Rajesh Gupta, University of California, San Diego

Energy is a precious societal resource, and increasingly rated for its quality or lack thereof as a contributor to greenhouse gases. Modern electrical energy systems operate at the intersection of technological advances in microelectronics, communications, and control. From individual components and systems such as computer systems to their aggregates and enclosures such as data centers and buildings, microelectronic advances in radios, processors, storage and networking are enabling low-cost and effective embedded sensing and its use in operational controls. In the context of energy distribution systems, this trend has led to popular visions of smart electrical grids that dynamically match generation, transmission, and storage for the most efficient and reliable usage of electromagnetic energy. This talk examines the technology components and their use methods that make an electrical grid efficient and reliable by increasing the visibility available to grid operators and end consumers alike. We show how microgrids, which are self-managed grids with local cogeneration capabilities, can be used as testing grounds for the prototyping and testing of smart grid technologies. Using the prototype of a microgrid at the campus of the University of California at San Diego, we present energy data that points to promising methods for operation of various types of buildings that leverage coordinated use of sensing, information processing, and building HVAC systems. Based on measurements and analyses, we show

that for the emerging class of mixed-use buildings that is, buildings with a non-trivial component of energy use by IT equipment -- significant possibilities exist to reduce total energy use from 10% to 30% based on effective duty-cycling of the IT and HVAC equipment, without affecting the comfort quality or availability of the building and compute resources.

Track 5 – E. Array Processing and Statistical Signal Processing

Session: WAb7 – MIMO Radar

Chair: ***Benjamin Friedlander,*** *University of California, Santa Cruz*

WA7b-1 10:15 AM

High Resolution Parameter Estimation for Ultra-Wideband MIMO Radar

Jussi Salmi, Aalto University; Seun Sangodoyin, Andreas Molisch, University of Southern California

The paper discusses modeling and parameter estimation for UWB radar, with the application of indoor localization and surveillance in mind. The goal is to identify small-scale movement, such as human breathing, based on UWB radar measurement and analysis of the estimated propagation path components. A wideband antenna array model is derived, and a modification of the RIMAX algorithm is employed for estimating the propagation path parameters. The results indicate that the resolution of the obtained delay estimates can be much better than the often assumed inverse of the measurement bandwidth.

WA7b-2 10:40 AM

Quadrature Slow-Time MIMO Radar with Experimental Results

Jason Yu, Jeffrey Krolik, Duke University

Multiple-input multiple-output (MIMO) radar requires orthogonal waveforms to be transmitted on each transmit element or subarray. One practical problem that arises is the complexity and expense of the hardware needed to implement MIMO radar. Slow-time MIMO waveforms, often referred to as SLO-MO waveforms, allow the implementation of MIMO radar without any hardware modifications to the receiver. However, generation of the SLO-MO MIMO waveforms on the transmitter typically requires a multichannel arbitrary waveform generator, variable RF phase shifters, or using multiple carrier frequencies. All of these options can be expensive, especially at microwave frequencies. This paper describes a technique to generate SLO-MO MIMO waveforms using low-cost passive frequency mixers with existing (non-MIMO) radar architectures. Passive frequency mixers are inexpensive and easy to add inline to existing radar transmitters. This paper presents the concept of double-sideband and quadrature MIMO slow-time radar, along with experimental results from the Duke S-band radar testbed.

WA7b-3 11:05 AM

The Applicability of GMTI MIMO Radar

Michael Zatman, QinetiQ North America

MIMO Radar has been proposed as a technique for improving the Minimum Detectable Velocity (MDV) performance of airborne radar systems. However, the increased pulse repetition frequency associated with waveform multiplexing techniques such as Doppler Division Multiple Access increases the amount of range ambiguous clutter the radar must suppress, reducing the amount of clutter-free Doppler space available to detect targets and often degrading MDV performance. This paper evaluates this effect and quantifies the conditions where MIMO radar is advantageous, and those where it is not. Results show that this form of MIMO radar is often a disadvantage at higher frequencies.

WA7b-4 11:30 AM

MIMO-VSAR: A High Resolution Radar System for Imaging Moving Scenes

Benjamin Friedlander, University of California, San Diego

The Velocity SAR (VSAR) is a SAR-based sensor system for high resolution imaging of moving scenes such as the ocean surface. VSAR utilizes data collected by a multi-element SAR system, to extract information not only about the radar reflectivity of the observed area, but also about the radial velocity of the scatterers in each pixel. This is accomplished by making use of the phase information contained in multiple SAR images, and not just the magnitude information as in conventional SAR. Using this velocity information, the VSAR is able to compensate for the velocity distortion inherent in conventional SAR. MIMO-VSAR is a variation of VSAR employing multiple transmit antennas using orthogonal waveforms. It is shown that MIMO-VSAR significantly extends the unambiguous velocity estimation range and relaxes the design constraints of the VSAR system.

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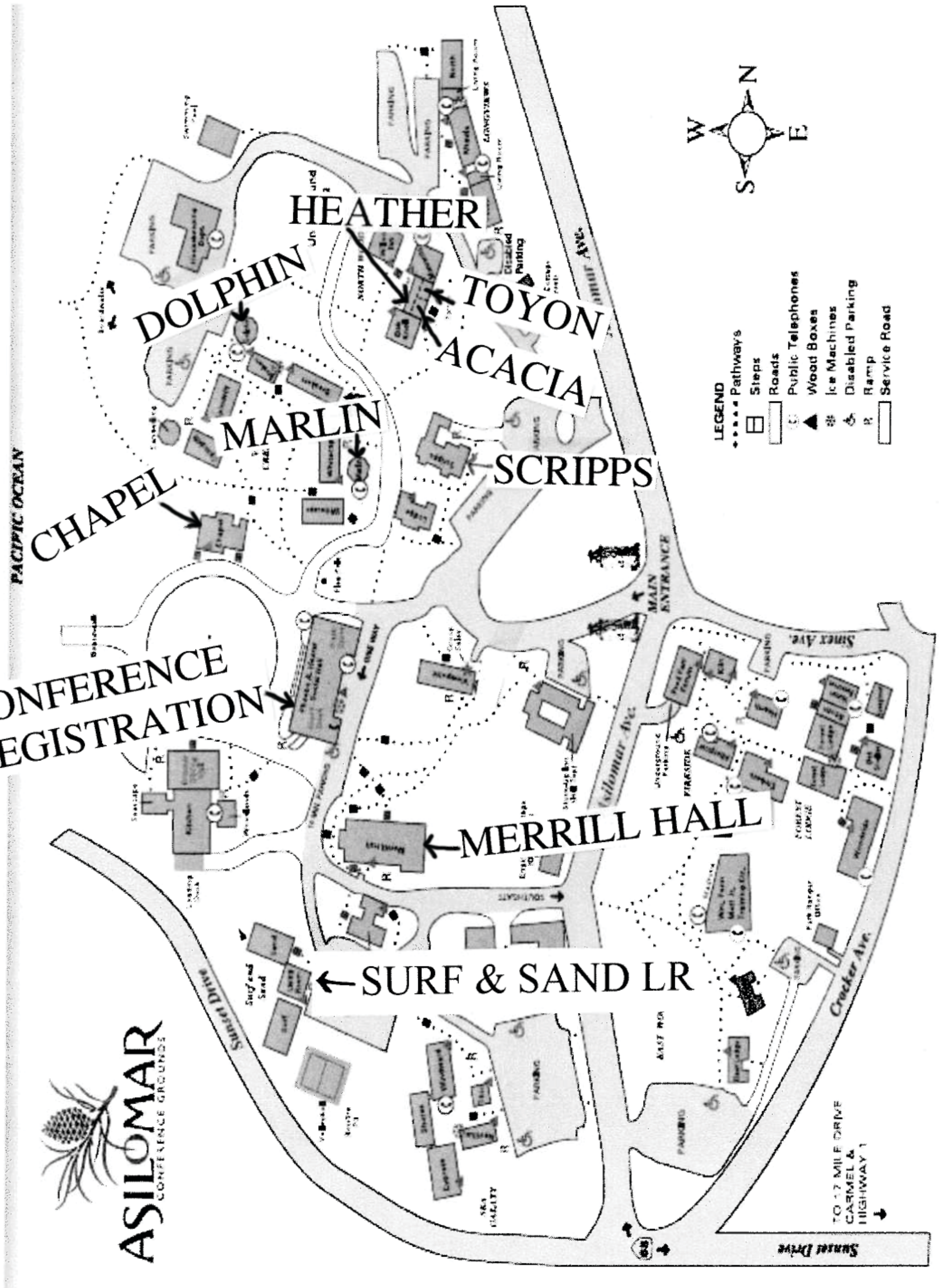
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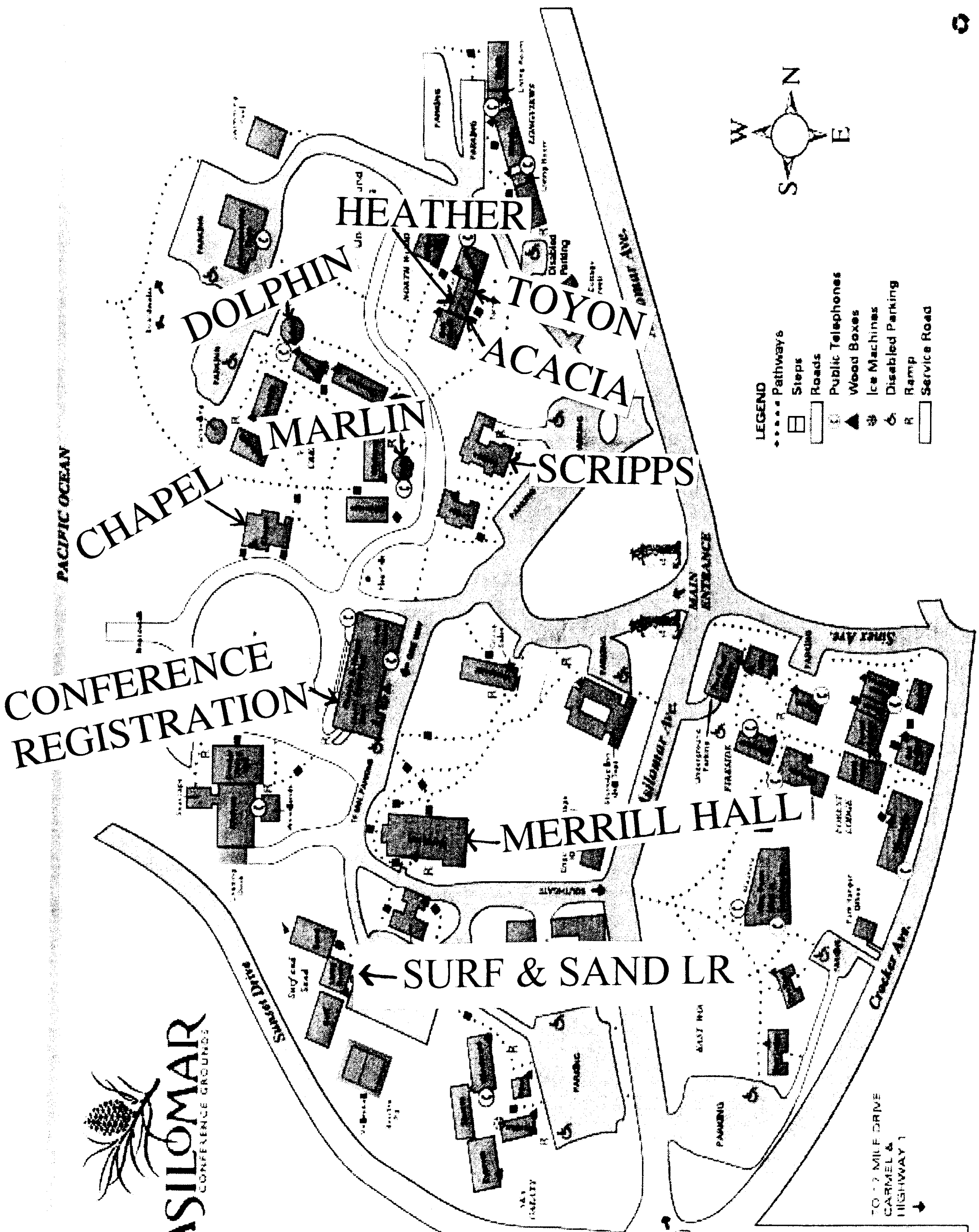
SUN PM	WELCOMING RECEPTION AT ASILOMAR 7:00-900 PM – MERRIL HALL							
Meeting Rooms	HEATHER	TOYON	ACACIA	DOLPHIN	SURF & SAND LR	SCRIPPS	MARLIN	MERRILL HALL
MON AM 8:15-9:55 & 10:15-12:00 **[MA]**	CONFERENCE PLENARY SESSION – CHAPEL HALL **Plenary Speaker – Dr. Ronald W. Schafer** *A Celebration of DSP Technology* - **MA8a 8:15-9:55**							
	MA1b Tensors Methods in Signal Processing	MA2b MIMO Interference Networks	MA3b Security in Wireless Networks	MA4b New Trends in Sequential System Identification	MA5b Biomotivated Recognition and Detection	MA6b Computer Arithmetic I	MA7b Bio. Models of Speech Perception & Applications in Automatic Speech Processing	**Poster Session – MA8b** **10:15-12:00** 8b1 – Comm. Systems I 8b2 – Selected Topics in Image Processing 8b3 – Applications of Compressive Sensing
MON PM 1:30-3:10 & 3:30-5:10 [**MP**]	MP1a Interference Channels	MP2a MIMO Secrecy	MP3a New Trends in Information Theory and Networks	MP4a Biomedical Image Analysis	MP5 Statistical Signal Processing for Complex Systems	MP6 Comm. Processors and Accelerators	MP7a Video Compression	**Poster Sessions – MP8a** **1:30-3:00** 8a1 – Comm. Systems II 8a2 – Speech Enhancement 8a3 – Topics in Speech & Audio 8a4 – Adaptive SP in Comm. 8a5 – Array-based Estimation
	MP1b Trends for Future Wireless Systems	MP2b MIMO Relays	MP3b Learning and Optimization in Dynamic Networks	MP4b Advances in Adaptive Algorithms			MP7b Advances in Keyword Spotting	**No poster sessions** **3:30 – 5:00**
TUE AM 8:15-9:55 & 10:15-12:00 **[TA]**	TA1a Network Error Correction and Physical Layer Security	TA2a Signal Processing for Comm. Receivers	TA3a Recursive Reconstruction of Sparse Sequences	TA4a Shape and Time in Biomedical Images	TA5 Compressive Sensing	TA6a Reconfigurable Architectures, Algorithms and Applications	TA7 Image and Video Enhancement	**Poster Sessions – TA8a** **8:15-9:55** 8a1 - Coop. and Cognitive Transmission in Multi-Antenna Networks I 8a2 – Cognitive Networking 8a3 – Adaptive Signal Proc. Theory & Applications
	TA1b Coding	TA2b Comm. Under Doppler Spread	TA3b Self-Organizing Networks: Architectures, Protocols and Algorithms	TA4b Mathematical Methods for Biomedical Signals and Images		TA6b Array Processing and Beamforming		**Poster Sessions – TA8b** **10:15-12:00** 8b1- Coop. and Cognitive Transmission in Multi-Antenna Networks II 8b2 – Architectures, Implement, & Tools I 8b3 – Architectures, Implement. & Tools II
Meeting Rooms	HEATHER	TOYON	ACACIA	DOLPHIN	SURF & SAND LR	SCRIPPS	MARLIN	MERRILL HALL
TUE PM 1:30-3:10 & 3:30-5:10 **[TP]**	TP1a Advances in Multi-hop and Distributed Wireless Transmission	TP2a MIMO Underwater Acoustic Comm.	TP3a Non-Stationary Processing of Environments	TP4a Modeling for Biomedical Imaging	TP5a Statistical Signal Processing for Neural Signals	TP6a Computer Arithmetic II	TP7a Microphone Array Processing for Speech Applications I	**Poster Sessions – TP8a** **1:30-3:00** 8a1 - Low Complexity Implement.& Receivers 8a2 - Detection & Estimation in Networks 8a3 – Techniques in Networking & Comm.
	TP1b Wireless Comm.	TP2b MIMO for Ad Hoc Networks	TP3b Network Information Theory	TP4b Adaptive Filters – Theory and Applications	TP5b Integrated Multimodal Sensing	TP6b Computer Arithmetic III	TP7b Microphone Array Processing for Speech Applications II	**Poster Sessions - TP8b** **3:30-5:00** 8b1-Scheduling, Relaying and Routing 8b2 – Statistical and Adaptive Signal Processing 8b3 – Biomedical Signals and Images
WED AM 8:15-9:55 & 10:15-12:00 **[WA]**	WA1a Cooperative Comm.	WA2a Interference Management I	WA3a Sensor Networks	WA4 Advances on Adaptive Filtering and Applications	WA5 Statistical Signal Processing	WA6a Estimation and Detection	WA7a Sparse Representations in Image Processing	No poster sessions Wed AM
	WA1b Comm. Theory	WA2b Interference Management II	WA3b Multiuser Beamforming and Interference Channels			WA6b SOC Architectures and Applications	WA7b MMO Radar	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