Issue 7

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APSIPA NEWSLETTER

Asia-Pacific Signal and Information Processing Association Newsletter

Issue Highlights

In APSIPA Newsletter issue-7, we have many columns that are quite interesting to share with the APSIPA community. We share the good news regarding the APSIPA Transactions on Signal and Information Processing (ATSIP). ATSIP has been approved for indexing under Scopus which is an outstanding recognition to this young journal. The actual indexing will take place in the near future. This would be a stepping stone toward the Web of Science indexing. We thus encourage all APSIPA members to submit their top publications to ATSIP. Information regarding APSIPA next conference in Siem Reap on the 9th of December is also depicted in the conference flyer. We introduce exciting articles showing the research activity of APSIPA members. The first two articles are extracts from the 2013 APSIPA conference best paper awards. An article about Nobel Prize winning papers being rejected by elite journals to prosper disappointed researchers can be read in this issue too. An interesting column is talking about APSIPA Online Education as a new paradigm to go with the current wave of on-line education and MOOCs. Book review and PhD thesis abstract with a link to download are also columned in this issue.

In this regard, we encourage all APSIPA members to send over their contributions to our newsletter to bloom knowledge and strengthen communications amongst our community. Contributions should not be more than three pages, with font size 10, including images. The contributions could be (but not limited) articles, announcements, ideas, or notes on:

1. Review on a recently published book, paper, or monograph.
2. Interesting technology, research line, software, or application.
3. Motivating scientific facts, progress, or development.
4. Review or tutorial about a research area.
5. Snapshots from pioneers in science and technology.
6. News and announcements especially for interns, scholarships, PG studies, fellowships, postdocs, funding, or jobs.
7. Lectures in signal and information technology (URL link to the video/ presentation slides are required)
8. Theses' abstracts passed examination (URL Link to the full theses are required).

In this issue

Granting institution, student and supervisor names, and contact emails are required to provide.

9. Pedagogical practices and teaching strategies to create improved learning environments.
10. Materials may participate in achieving APSIPA objectives.

All submissions should be sent to (w.abdulla@auckland.ac.nz) with a subject heading 'APSIPA Newsletter'. We reserve the right to carry on minor editing to all submissions to meet our editorial procedure. Copyrights of material remain with the original author who grants us the right to make it available on APSIPA website.

Awaiting your contributions to the next issue of APSIPA Newsletter

Waleed Abdulla
APSIPA Newsletter Editor-in-Chief

APSIPA Online Education – A New Paradigm

Eliathamby Ambikairajah, Member of APSIPA Advisory Board

One of the ways by which APSIPA serves its research communities is through the APSIPA Distinguished Lecturer Program where it organises lectures delivered by distinguished experts. These experts also serve as ambassadors for APSIPA to promote its image and encourage new membership. While the program has successfully organised lectures over the past 3 years, due to their nature, these lectures are one-off events and consequently their reach is small. Additional avenues of reaching out to the communities should be considered and the available technology for online education can be utilised for this purpose. A series of online tutorials on topics related to a selected research area, delivered by distinguished experts, *recorded offline* and hosted by APSIPA may serve as a valuable educational resource for its communities. For example, in the area of speaker verification, individual tutorials on feature extraction, feature normalisation, model training, model normalisation, score normalisation and system fusion can be delivered by APSIPA experts in those fields and recorded offline. Members of the 'APSIPA Friend Labs' provide an ideal pool of experts to draw from to deliver these tutorials. Given the number of researchers working in this field, and the number of new PhD students embarking on this research, such a series of high calibre, well-crafted and presented tutorials may serve as an invaluable resource for these early researchers. With new techniques emerging in this field, the 'APSIPA speaker verification tutorial series' can grow by adding new tutorials to keep pace with the state-of-the-art. Similar tutorial series for other research areas such as speech recognition, language identification, video coding, machine learning, etc. together will form the APSIPA Open Online Tutorial repository. A valuable online resource such as this will be unique to APSIPA among all professional bodies and has the potential to attract a significant amount of international attention. Further, in the future some of these tutorial series may be developed into MOOCs run by APSIPA.

APSIPA Membership Board

The APSIPA Membership Board (AMB) was set up in February 2014. The AMB is led by APSIPA VP-Member Relations and Development, Professor Kenneth Lam, with the following members:

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The AMB members will serve a renewable 2-year term, with a maximum of two consecutive terms. The AMB will actively promote APSIPA to all researchers and academics in the fields of signal and information processing, and also organize APSIPA activities for the benefit of APSIPA members and e-members.

CAMBRIDGE JOURNALS

APSIPA TRANSACTIONS ON SIGNAL AND INFORMATION PROCESSING

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Now accepted for indexing by Scopus

On behalf of the Editorial Board of *APSIPA Transactions on Signal and Information* (*Transactions*), I am very pleased to announce that the journal is now indexed by Scopus.

Scopus is one of the premier indexing services and an important step towards having *Transactions* fully indexed. Scopus joins Google Scholar and other services that already list the journal.

Since the first papers appeared, less than 2 years ago, *Transactions* has steadily grown in visibility within the broad Signal and Information Processing research community. Recent highlights include:

- **An overview on video forensics**, Simone Milani, Marco Fontani, Paolo Bestagini, Mauro Barni et al.
- **Recent advances on active noise control: open issues and innovative applications**, Yoshinobu Kajikawa, Woon-Seng Gan and Sen M. Kuo
- **A tutorial survey of architectures, algorithms and applications for deep learning**, Li Deng
- **Survey on securing data storage in the cloud**, Chun-Ting Huang, Lei Huang, Zhongyuan Qin, Hang Yuan et al.

Editor-in-Chief

Antonio Ortega (University of Southern California, USA)

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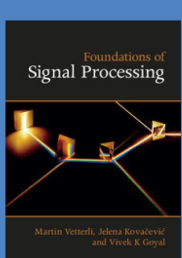

Coming soon
Cambridge University Press is publishing a new book co-authored by *Transactions* Advisory Board member Martin Vetterli.

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Sum-Rate Maximization and Energy-Cost Minimization for Renewable Energy Empowered Base-Stations using Zero-Forcing Beamforming

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I. Introduction

Small cell networks [1] have emerged as a promising solution to meet the increasing demand for high data-rate wireless communications in recent years. The reduced distance between base-stations (BSs) and users allow BSs to operate under lower power, enabling the use of renewable energy. However, the uncertainty of renewable energy arrivals and the limitation on the battery storage capacity pose additional constraints to the downlink transmission. A redesign of the transmit signals and power control policies is necessary for efficient usage of the renewable energy and for achieving quality-of-service (QoS) guarantees under these constraints.

Downlink beamforming and power control policies have been studied extensively in the literature based on different design criteria [2]–[5]. Among these schemes, zero-forcing (ZF) beamforming is considered as one of the most popular and practical multiuser beamforming schemes due to its simplicity and ability to eliminate interuser interference in the downlink of a multiuser multiple-input single-output (MISO) wireless system. In fact, this scheme is known to achieve the optimal degrees-of-freedom performance [3] and asymptotically optimal sum-rate performance as the number of users goes to infinity [4]. These techniques can also be extended to multiple-input multiple-output (MIMO) as well. However, the above works do not consider the impact of renewable energy constraints in the design.

The need for the redesign of transmission policies in renewable energy enabled systems were demonstrated recently in [6]-[8]. In [6], offline and online power control policies were derived for a single-input single-output point-to-point channel. The directional water-filling algorithm was proposed and shown to be optimal in terms of maximizing the throughput by a deadline subject to energy causality and battery storage constraints. These studies were extended to the K-user AWGN broadcast channel in [7]. Moreover, in [8], the authors considered random data arrivals and discussed a grid power minimization problem. The problem is shown to be a dual problem of the throughput maximization problem when all data is available prior to transmission. Different from these works, we consider the multiuser downlink beamforming scenario with a multi-antenna BS. Under ZF constraints, we determine the joint beamforming and power control policy in this scenario.

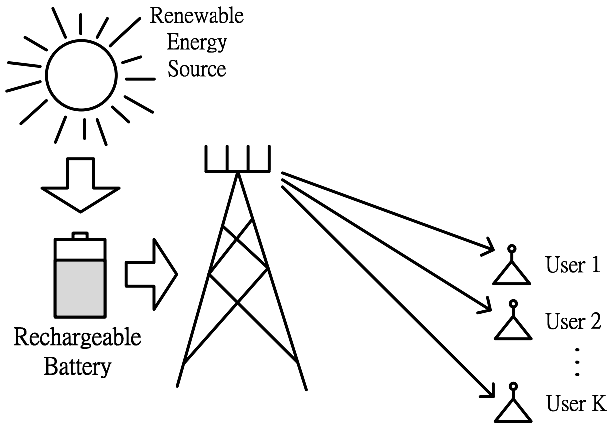

Fig. 1. Small cell BS with renewable energy source.

II. Main Objective

The main objective of this work is devise ZF beamforming and power control policies for efficient usage of renewable energy at a small cell BS. The BS is equipped with multiple antennas and is supported by renewable energy through a rechargeable battery. The policies are derived based on two optimizing criterion: the maximum sum-rate criterion and the minimum energy-cost criterion.

In the sum-rate maximization problem, the BS is assumed to be supported only by renewable energy and the goal is to maximize the sum-rate of all users and by the deadline. In the energy-cost minimization problem, the BS is assumed to be supported by both renewable energy and grid energy, and the goal is to minimize the total cost of purchasing grid energy over time. Due to the usage of renewable energy, the designs are subject to energy causality and battery storage constraints, i.e., the constraint that energy cannot be used before it arrives and the constraint that the stored energy cannot exceed the maximum storage capacity of the rechargeable battery.

III. System Model and Problem Formulation

Consider a downlink wireless cellular system with a multi-antenna BS transmitting to K single-antenna users, as illustrated in Fig. 1. The BS is equipped with M antennas and is supported by renewable energy through a rechargeable battery. Here, we consider a time-slotted system and assume that the channels remain constant within each time slot. All K users are served simultaneously by the BS in each time slot. In this case, it is necessary to have $M \geq K$ in order to completely eliminate inter-user interference.

Let $s_i[t]$ be the signal intended for user i in time slot t and let $w_i[t]$ be the associated $M \times 1$ beamforming vector. Assume that $s_i[t]$ is Gaussian with zero mean and unit variance, and is independent and identically distributed (i.i.d.) for all i and t. The signal transmitted by the BS in time slot t can then be expressed as $\mathbf{x}[t] = \sum_{i=1}^{K} \mathbf{w}_i[t] s_i[t]$, and the power of the signal is given as $\mathbf{E}[\|\mathbf{x}[t]\|^2] = \sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2$. The received signal at user i is given by

$$y_i[t] = \mathbf{h}_i[t]^H \mathbf{w}_i[t] s_i[t] + \mathbf{h}_i[t]^H \sum_{j \neq i} \mathbf{w}_j[t] s_j[t] + n_i[t],$$

where $\mathbf{h}_i[t]$ is the $M \times 1$ channel vector between the BS and user i and $n_i[t]$ is the additive white Gaussian noise (AWGN) with zero mean and variance σ_n^2. The second term of the received signal contains the interference signal from other signal for other users. To eliminate the inter-user interference, the zero-forcing (ZF) beamforming vectors are chosen such that $\mathbf{h}_i[t]^H \mathbf{w}_j[t] = 0$ for all $i \neq j$. The received signal-to-noise ratio (SNR) at user i is given as $\mathrm{SNR}_i[t] = |\mathbf{h}_i[t]^H \mathbf{w}_i[t]|^2 / \sigma_n^2$ and the achievable throughput by a deadline T can be expressed as

$$\sum_{t=1}^{T} \sum_{i=1}^{K} \log_2(1 + \mathrm{SNR}_i[t]). \tag{1}$$

Due to renewable energy usage at the BS, the total transmission power in each time slot is subject to energy causality constraint and battery capacity constraints. Let $\epsilon[t]$ be the amount of energy that can be harvested by the BS at the beginning of time slot t. To model the effect that the energy arrival may not be stored completely into the battery, an additional variable $B_{\mathrm{in}}[t]$ is introduced to represent the energy actually stored into the battery at time t $(B_{\mathrm{in}}[t] \leq \epsilon[t], \forall t)$. Without considering grid energy usage, the energy causality and the battery storage constraints can be written as

$$\sum_{t=1}^{\ell} \sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2 \leq \sum_{t=1}^{\ell} B_{\mathrm{in}}[t] \ , \tag{2}$$

for $\ell = 1, \cdots, T$, and

$$\sum_{t=1}^{\ell+1} B_{\mathrm{in}}[t] - \sum_{t=1}^{\ell} \sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2 \leq B_{\max} \ , \tag{3}$$

for $\ell = 1, \cdots, T-1$.

In the sum-rate maximization problem, the BS is assumed to be supported only by renewable energy and the goal is to maximize the sum-rate over all users subject to energy causality, battery storage, and ZF constraints. The problem can be formulated as:

$$\begin{aligned} \max_{\mathbf{w}_i[t], B_{\mathrm{in}}[t], \forall i, t} \ & \sum_{t=1}^{T} \sum_{i=1}^{K} \log_2\left(1 + \frac{|\mathbf{h}_i[t]^H \mathbf{w}_i[t]|^2}{\sigma_n^2}\right) \\ \text{subject to} \ & (2), \quad (3), \\ & \mathbf{h}_i[t]^H \mathbf{w}_j[t] = 0, \quad \forall i \neq j, \\ & 0 \leq B_{\mathrm{in}}[t] \leq \epsilon[t], \ \forall i, \forall \ell, \forall t. \end{aligned} \tag{4}$$

In the energy-cost minimization problem, the BS is assumed to be supported by both renewable and grid energy, and the goal is to minimize the total cost of grid energy usage subject to QoS constraints at all users as well as energy causality and battery storage constraints. We assume that the energy consumed from the grid is not stored into the battery for later use. The cost is computed according to a time-varying convex cost function f_t so that the cost of purchasing energy $B_{\mathrm{grid}}[t]$ in time slot t is $f_t(B_{\mathrm{grid}}[t])$. The QoS constraint at user $i$is given by $\mathrm{SNR}_i[t] \geq \gamma_i$, where γ_i is the target SNR. The off-line energy-cost minimization problem can be formulated as:

$$\begin{aligned} \min_{\mathbf{w}_i[t], B_{\mathrm{grid}}[t], B_{\mathrm{in}}[t]} \ & \sum_{t=1}^{T} f_t\left(B_{\mathrm{grid}}[t]\right) \\ \text{subject to} \ & \\ & \sum_{t=1}^{\ell} \left(\sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2 - B_{\mathrm{grid}}[t] \right) \leq \sum_{t=1}^{\ell} B_{\mathrm{in}}[t], \\ & \sum_{t=1}^{\ell+1} B_{\mathrm{in}}[t] - \sum_{t=1}^{\ell} \left(\sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2 - B_{\mathrm{grid}}[t] \right) \leq B_{\max}, \\ & \frac{|\mathbf{h}_i[t]^H \mathbf{w}_i[t]|^2}{\sigma_n^2} \geq \gamma_i, \ \forall i, \quad \mathbf{h}_i[t]^H \mathbf{w}_j[t] = 0, \quad \forall i \neq j, \\ & 0 \leq B_{\mathrm{in}}[t] \leq \epsilon[t], \quad 0 \leq B_{\mathrm{grid}}[t] \leq \sum_{i=1}^{K} \|\mathbf{w}_i[t]\|^2, \ \forall i, \forall \ell, \forall t. \end{aligned} \tag{5}$$

Notice that the problems are non-convex in their original forms. However, they can be transformed into convex optimization problems and can be solved efficiently using off the-shelf interior point solvers, such as CVX [9].

IV. Sum-Rate Maximization Problem

In this section, we derive the optimal beamforming and power control policies under the sum-rate criterion.

To do so, we first introduce the following lemma:

Lemma 1: *The choice of renewable energy storage*

$$B_{\text{in}}^*[t] = \min\{\epsilon[t], B_{\max}\}, \quad \text{for } t = 1, \ldots, T, \tag{6}$$

is optimal for the sum-rate maximization problem.

This lemma shows that all the renewable energy should be stored into the battery for the transmission at each time slot. If the arrival energy exceed the capacity of the battery size, then the amount of the stored energy should be equal to $B_{\max}$.

Given the above lemma, what remains to be determined are the beamforming vectors $\mathbf{w}_i[t]$. Let us define the transmit covariance matrix corresponding to the signal for user i as $\mathbf{Q}_i[t] = \mathbf{w}_i[t]\mathbf{w}_i^H[t]$. The covariance matrix is a positive semidefinite matrix with rank equal to 1. The ZF constraint can be represented as $\mathbf{h}_i^H[t]\mathbf{Q}_j[t]\mathbf{h}_i[t] = 0$ for $i \neq j$. By substituting the optimal value of $B_{\text{in}}[t]$ and the transmission covariance matrix into the above problem, the optimal structure of the covariance matrix can be obtained as shown in [10], and we have the following lemma.

Lemma 2 ([10]): *The ZF constraint yields the optimal structure of* $\mathbf{Q}_i[t]$ *as*

$$\mathbf{Q}_i^*[t] = \tilde{\mathbf{V}}_i[t]\tilde{\mathbf{q}}_i^*[t]\lambda_i[t]\tilde{\mathbf{q}}_i^*[t]^H\tilde{\mathbf{V}}_i[t]^H, \forall i, t, \tag{7}$$

where $\tilde{\mathbf{V}}_i[t]$ *is an* M *-by-* $(M-K+1)$ *matrix constructed by the null space of the interfering channels,* $\lambda_i[t]$ *is the power allocated to user* i *and* $\tilde{\mathbf{q}}_i^*[t] = \tilde{\mathbf{V}}_i[t]^H\mathbf{h}_i[t]/\|\tilde{\mathbf{V}}_i[t]^H\mathbf{h}_i[t]\|$ *is the projection of the channel direction* $\mathbf{h}_i[t]$ *onto the null space of all interference channels.*

By the above lemma, the ZF constraint is satisfied with the choice of $\mathbf{Q}_i[t]$. Here, we have $\text{tr}(\mathbf{Q}_i[t]) = \text{tr}(\tilde{\mathbf{Q}}_i[t]) = \lambda_i[t]$ and, thus, the power constraints on $\mathbf{Q}_i[t]$ in above problems can be replaced by $\lambda_i[t]$.

Given the beamforming direction, the problem in (4) reduces to the power control problem:

$$\begin{aligned}
\max_{\lambda_i[t], \forall i, \forall t} \ & \sum_{t=1}^{T}\sum_{i=1}^{K} \log_2\left(1 + \frac{\lambda_i[t]\|\tilde{\mathbf{V}}_i[t]^H\mathbf{h}_i[t]\|^2}{\sigma_n^2}\right) \\
\text{subject to} \ & \sum_{t=1}^{\ell}\sum_{i=1}^{K}\lambda_i[t] \leq \sum_{t=1}^{\ell} B_{\text{in}}^*[t], \\
& \sum_{t=1}^{\ell+1} B_{\text{in}}^*[t] - \sum_{t=1}^{\ell}\sum_{i=1}^{K}\lambda_i[t] \leq B_{\max}, \\
& \lambda_i[t] \geq 0, \quad \forall i, \forall \ell, \forall t.
\end{aligned} \tag{8}$$

The problem is convex and can be solved by considering its Lagrangian dual problem and by applying the KKT optimality conditions. The optimal power allocation can be found as

$$\lambda_i^*[t] = \left(\frac{1}{\tau_t} - \frac{1}{\beta_i[t]}\right)^+, \forall i, t, \tag{9}$$

In the above, $\beta_i[t] = \frac{\|\tilde{\mathbf{V}}_i[t]^H\mathbf{h}_i[t]\|^2}{\sigma_n^2}$ can be viewed as the equivalent channel gain and the parameter τ_t is defined as $\tau_t = \sum_{\ell=t}^{T}\zeta_\ell - \sum_{\ell=t}^{T-1}\mu_\ell$, where $\zeta_\ell \geq 0$ and $\mu_\ell \geq 0$ are the Lagrangian multipliers. Notice that the power allocation among users in equivalent to a traditional water-filling solution in time slot t, where the water-level equals to $1/\tau_t$. It means that the power allocation can be performed in two step algorithm by first determining the water-level in each time slot and then by computing the water-filling solution over different users. According to the analysis of KKT conditions, we use the proposed directional water-filling algorithm in [6] to find the optimal power allocation.

V. Energy-Cost Minimization Problem

For the energy-cost minimization problem, we do not choose to replace $B_{\text{in}}[t]$ with $B_{\text{in}}^*[t]$ given in Lemma 1, even though we can show that there is no loss of optimality in doing so. Let us again define the transmit covariance matrix corresponding to the signal for user i as $\mathbf{Q}_i[t] = \mathbf{w}_i[t]\mathbf{w}_i^H[t]$. By Lemma 2, the optimal transmit covariance is given by $\mathbf{Q}_i^*[t] = \tilde{\mathbf{V}}_i[t]\tilde{\mathbf{q}}_i[t]\lambda_i[t]\tilde{\mathbf{q}}_i[t]^H\tilde{\mathbf{V}}_i[t]^H$. Notice that the beamforming direction $\tilde{\mathbf{q}}_i[t]$ appears only in the QoS constraints. To minimize the total energy cost, one should choose $\tilde{\mathbf{q}}_i[t]$ such that the SNR of user i is as large as possible, and the energy required to achieve the SNR constraints can be minimized. Therefore, the optimal $\tilde{\mathbf{q}}_i^*[t]$ is given the same as the result in Lemma 2, and the optimal value of $\lambda_i[t]$ should be chosen such that the SNR constraint is met with equality. This is given by $\tilde{\lambda}_i^*[t] = \sigma_n^2\gamma_i/|\mathbf{h}_i^H[t]\tilde{\mathbf{V}}_i[t]\tilde{\mathbf{q}}_i^*[t]|^2$. With the optimal beamforming direction $\tilde{\mathbf{q}}_i^*[t]$, the off-line problem (5) reduces to the following optimization problem:

$$\min_{B_{\rm grid}[t],B_{\rm in}[t],\forall t} \sum_{t=1}^{T} f_t(B_{\rm grid}[t]) \quad (10)$$
$$\text{s. t.} \sum_{t=1}^{\ell}\left(\sum_{i=1}^{K}\tilde{\lambda}_i^*[t]-B_{\rm grid}[t]\right)\le\sum_{t=1}^{\ell}B_{\rm in}[t],$$
$$\sum_{t=1}^{\ell+1}B_{\rm in}[t]-\sum_{t=1}^{\ell}\left(\sum_{i=1}^{K}\tilde{\lambda}_i^*[t]-B_{\rm grid}[t]\right)\le B_{\max},$$
$$0\le B_{\rm grid}[t]\le\sum_{k=1}^{K}\tilde{\lambda}_i^*[t],\quad 0\le B_{\rm in}[t]\le\epsilon[t],\forall\ell,\forall t.$$

In the above, only the grid energy $B_{\rm grid}[t]$ and the amount of stored renewable energy $B_{\rm in}[t]$ remains to be determined. The above problem is a convex optimization problem which can be solved efficiently using interior point solvers. We should notice that, by considering a convex cost function, the BS is penalized more for consuming a large amount of grid energy at any given time. Hence, it is more desirable to spread the grid energy consumption over time. Even though we assume that grid energy is not stored into the battery, this can still be achieved by reallocating the renewable energy usage over time. Moreover, when the cost function varies over time, it is also desirable to utilize less renewable energy (when the grid energy cost is low), and leave it for use in later time slots (when grid energy cost is high).

VI. Conclusion

Both offline and online ZF beamforming and power control policies were presented for the downlink of renewable-energy enabled multi-antenna BSs. Two design problems were considered, i.e., the sum-rate maximization problem and energy-cost minimization problem. In the SRMax problem, the optimal ZF beamformer was derived explicitly and was shown to be consistent with that found for systems without renewable energy constraints. The power control problem was then shown to be convex and to result in the directional water-filling algorithm. In the ECMin problem, the cost of consuming grid energy was minimized by allowing BSs to reduce the use of renewable energy in time slots with lower energy cost and store it in the battery for use in time slots with higher energy cost.

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Musical-Noise-Free Noise Reduction Theory Based on Higher-Order Statistics

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I. Introduction

In recent studies, many applications of hands-free speech communication systems have been investigated, for which noise reduction is a problem requiring urgent attention. Spectral subtraction is a commonly used noise reduction method that has high noise reduction performance [1]. However, in this method, artificial distortion, so-called *musical noise*, arises owing to nonlinear signal processing, leading to a serious deterioration of sound quality.

To achieve high-quality noise reduction with low musical noise, an *iterative spectral subtraction* method has been proposed [2]. This method is performed through signal processing in which *weak* spectral subtraction processes are recursively applied to the input signal. Also, Inoue, et al., have reported the very interesting phenomenon that this method with appropriate parameters gives *equilibrium* behavior in the growth of higher-order statistics with increasing number of iterations [3]. This means that almost no musical noise is generated even with high noise reduction, which is one of the most desirable properties of single-channel nonlinear noise reduction methods. However, the existence of the musical-noise-free state has been discovered only experimentally, and there have been no theoretical studies on it.

Therefore, in this paper, I review my current study on the above mentioned problem [4]. First, I theoretically derive a closed-form solution of the internal parameters that satisfy the musical-noise-free condition by analysis based on higher-order statistics. It is clarified that finding a fixed point in the kurtosis of noise spectra enables the reproduction of the musical-noise-free state. Also, comparative experiments with commonly used noise reduction methods show the efficacy of the proposed method via subjective evaluation.

II. Related Works

A. Conventional non-iterative spectral subtraction

We apply short-time Fourier analysis to the observed signal, which is a mixture of target speech and noise, to obtain the time-frequency signal. We formulate conventional *non-iterative spectral subtraction* [1] in the time-frequency domain as follows:

$$y(f,\tau)=\begin{cases}\sqrt{|x(f,\tau)|^2-\beta\mathrm{E}[|N|^2]}e^{j\arg(x(f,\tau))} \\ \quad (\text{where} \quad |x(f,\tau)|^2-\beta\mathrm{E}[|N|^2]>0), \\ \eta x(f,\tau) \quad\quad (\text{otherwise}),\end{cases} \tag{1}$$

where $y(f,\tau)$ is the enhanced target speech signal, $x(f,\tau)$ is the observed signal, f denotes the frequency subband, τ is the frame index, β is the oversubtraction parameter, and η is the flooring parameter. Here, $\mathrm{E}[|N|^2]$ is the expectation of the random variable $|N|^2$ corresponding to noise power spectra. In practice, we can approximate $\mathrm{E}[|N|^2]$ by averaging the observed noise power spectra $|n(f,\tau)|^2$ in the first K-sample frames, where we assume speech absence in this period; $\widehat{\mathrm{E}[|N|^2]} \approx \frac{1}{K}\sum_{\tau=1}^{K}|n(f,\tau)|^2$.

B. Modeling of input signal

In this paper, we assume that the input signal x in the power spectral domain is modeled using the gamma distribution as

$$P(x)=\frac{x^{\alpha-1}}{\Gamma(\alpha)\theta^{-\alpha}}\exp(-x/\theta), \tag{2}$$

where $x \geq 0, \alpha > 0$, and $\theta > 0$. Here, α is the shape parameter, θ is the scale parameter, and $\Gamma(\cdot)$ is the gamma function. The gamma distribution with $\alpha = 1$ corresponds to the chi-square distribution with 2 degrees of freedom.

C. Mathematical metric of musical noise generation via higher-order statistics for non-iterative spectral subtraction

In this study, we apply the *kurtosis ratio* to a *noise-only time-frequency period* of the subject signal for the assessment of musical noise [5]. This measure is defined as

$$\text{kurtosis ratio} = \mathrm{kurt}_{\mathrm{proc}}/\mathrm{kurt}_{\mathrm{org}}, \tag{3}$$

where $\mathrm{kurt}_{\mathrm{proc}}$ is the kurtosis of the processed signal and $\mathrm{kurt}_{\mathrm{org}}$ is the kurtosis of the observed signal. Kurtosis is defined as

$$\mathrm{kurt} = \mu_4/\mu_2^2, \tag{4}$$

where μ_m is the mth-order moment, given by

$$\mu_m=\int_0^\infty x^m P(x)dx, \tag{5}$$

and $P(x)$ is the probability density function (p.d.f.) of the random variable X. A kurtosis ratio of unity corresponds to no musical noise. This measure increases as the amount of generated musical noise increases.

The mth-order moment after spectral subtraction, μ_m^{SS}, is given by [3]

$$\mu_m^{\mathrm{SS}}=\theta^m\mathcal{M}(\alpha,\beta,\eta,m), \tag{6}$$

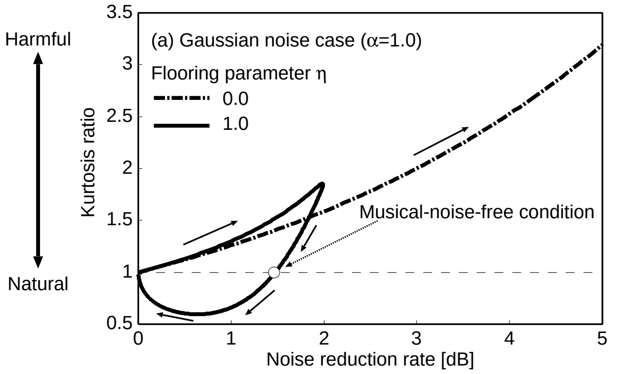

Fig. 1. NRR and kurtosis ratio obtained from theoretical analysis with increasing β. Note that *hysteresis loop* exists when $\eta = 1.0$.

where

$$\mathcal{M}(\alpha,\beta,\eta,m)=\sum_{l=0}^{m}(-\beta\alpha_0)^l\frac{\Gamma(m+1)\Gamma(\alpha_0+m-l,\beta\alpha_0)}{\Gamma(\alpha_0)\Gamma(l+1)\Gamma(m-l+1)} + \frac{\eta^{2m}}{\Gamma(\alpha)}\gamma(\alpha+m,\beta\alpha). \quad (7)$$

$\Gamma(b,a)$ and $\gamma(b,a)$ are the upper and lower incomplete gamma functions defined as $\Gamma(b,a) = \int_b^\infty t^{a-1}\exp(-t)dt$ and $\gamma(b,a) = \int_0^b t^{a-1}\exp(-t)dt$, respectively. From (4), (6), and (7), the kurtosis after spectral subtraction can be expressed as

$$\mathrm{kurt}(\alpha,\beta,\eta) = \mathcal{M}(\alpha,\beta,\eta,4)/\mathcal{M}^2(\alpha,\beta,\eta,2). \quad (8)$$

Using (3) and (8), we also express the kurtosis ratio as

$$\text{kurtosis ratio} = \frac{\mathcal{M}(\alpha,\beta,\eta,4)/\mathcal{M}^2(\alpha,\beta,\eta,2)}{\mathcal{M}(\alpha,0,0,4)/\mathcal{M}^2(\alpha,0,0,2)}. \quad (9)$$

Also, as a measure of noise reduction performance, the noise reduction rate (NRR), the output SNR minus the input SNR in dB, can be given in terms of a 1st-order moment as [3]

$$\mathrm{NRR} = 10\log_{10}\frac{\alpha}{\mathcal{M}(\alpha,\beta,\eta,1)}. \quad (10)$$

III. Theorem on Musical-Noise-Free Conditions

A. Overview

In this section, I introduce a *musical-noise-free* noise reduction theory that generates no musical noise. This method is based on iterative spectral subtraction and the analysis on the higher-order statistics discussed in the previous section. Figure 1 indicates the relation between NRR and kurtosis ratio in non-iterative spectral subtraction with various oversubtraction parameter β. From Fig. 1, we have found a *hysteresis loop* and we can confirm a musical-noise-free condition that is a point $\mathrm{NRR} > 0$ but $\mathrm{kurtosis ratio} = 1.0$ This means a special condition of non-iterative spectral subtraction can achieve noise reduction without musical noise generation. Then, we can achieve musical-noise-free noise reduction by iteratively performing spectral subtraction with parameters satisfying the musical-noise-free condition.

B. Derivation of musical-noise-free condition

To derive the musical-noise-free condition is equal to finding a fixed-point kurtosis condition. Although the parameters to be optimized are η and β, we hereafter derive the optimal η given a fixed β for ease of closed-form analysis.

First, we change (8) to

$$\mathrm{kurt}(\alpha_0,\beta,\eta)=\frac{\mathcal{S}(\alpha_0,\beta,4)+\eta^8\mathcal{F}(\alpha_0,\beta,4)}{\left(\mathcal{S}(\alpha_0,\beta,2)+\eta^4\mathcal{F}(\alpha_0,\beta,2)\right)^2}, \quad (11)$$

$$\mathcal{S}(\alpha_0,\beta,m)=\sum_{l=0}^{m}\frac{(-\beta\alpha_0)^l\Gamma(m+1)\Gamma(\alpha_0+m-l,\beta\alpha_0)}{\Gamma(\alpha_0)\Gamma(l+1)\Gamma(m-l+1)}, \quad (12)$$

$$\mathcal{F}(\alpha_0,\beta,m)=\frac{\gamma(\alpha_0+m,\beta\alpha_0)}{\Gamma(\alpha_0)}. \quad (13)$$

Next, the fixed-point kurtosis condition corresponds to the kurtosis being equal before and after spectral subtraction, thus

$$\frac{\mathcal{S}(\alpha_0,\beta,4)+\eta^8\mathcal{F}(\alpha_0,\beta,4)}{\left(\mathcal{S}(\alpha_0,\beta,2)+\eta^4\mathcal{F}(\alpha_0,\beta,2)\right)^2} = \frac{(\alpha_0+3)(\alpha_0+2)}{(\alpha_0+1)\alpha_0}. \quad (14)$$

Let $\mathcal{H} = \eta^4$, then (14) yields the following quadratic equation in $\mathcal{H}$.

$$\begin{aligned}&\left(\mathcal{F}(\alpha_0,\beta,4)(\alpha_0+1)\alpha_0-\mathcal{F}^2(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\right)\mathcal{H}^2\\&-2\mathcal{S}(\alpha_0,\beta,2)\mathcal{F}(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\,\mathcal{H}\\&+\mathcal{S}(\alpha_0,\beta,4)(\alpha_0+1)\alpha_0-\mathcal{S}^2(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)=0.\end{aligned} \quad (15)$$

Thus, we can derive a closed-form estimate of $\mathcal{H}$ from the given oversubtraction parameter as

$$\begin{aligned}\mathcal{H} =& \left\{\mathcal{F}(\alpha_0,\beta,4)(\alpha_0+1)\alpha_0-\mathcal{F}^2(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\right\}^{-1}\\&\Bigg[\mathcal{S}(\alpha_0,\beta,2)\mathcal{F}(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\\&\pm\Big[\left\{\mathcal{S}(\alpha_0,\beta,2)\mathcal{F}(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\right\}^2\\&-\left\{\mathcal{F}(\alpha_0,\beta,4)(\alpha_0+1)\alpha_0-\mathcal{F}^2(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\right\}\\&\left\{\mathcal{S}(\alpha_0,\beta,4)(\alpha_0+1)\alpha_0-\mathcal{S}^2(\alpha_0,\beta,2)(\alpha_0+3)(\alpha_0+2)\right\}\Big]^{\frac{1}{2}}\Bigg].\end{aligned} \quad (16)$$

Finally, $\eta = \mathcal{H}^{1/4}$ is the resultant flooring parameter that satisfies the fixed-point kurtosis condition.

Next, I reveal the range of the flooring parameter η that increases the NRR. From (10), the NRR growth condition is expressed as

$$\mathrm{NRR}=10\log_{10}\frac{\alpha_0}{\mathcal{S}(\alpha_0,\beta,1)+\eta^2\mathcal{F}(\alpha_0,\beta,1)} > 0. \quad (17)$$

Here, since $\eta > 0$, we can solve the inequality as

$$0 < \eta < \sqrt{\frac{\alpha_0-\mathcal{S}(\alpha_0,\beta,1)}{\mathcal{F}(\alpha_0,\beta,1)}}. \quad (18)$$

In summary, we can choose the parameters simultaneously satisfying the fixed-point kurtosis condition and NRR growth condition using (16) and (18).

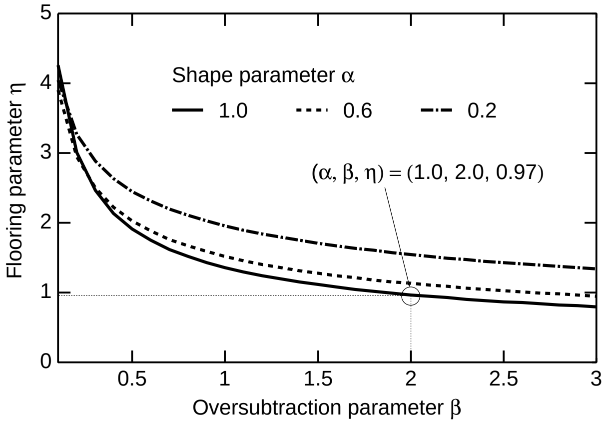

Fig. 2. Example of oversubtraction parameter β and flooring parameter η that satisfy musical-noise-free condition.

According to the previous analysis, we can calculate combinations of the oversubtraction parameter β and the flooring parameter η that satisfy the musical-noise-free condition under the three types of the shape parameter α. Figure 2 shows examples of traces, where the shape parameter α is set to 0.2, 0.6, and 1.0. It is worth mentioning that the specific setting $(\alpha, \beta, \eta) = (1.0, 2.0, 0.97)$ appears in Fig. 2, which was heuristically discovered in [3], but this theory can provide more wide-ranging solutions.

C. Subjective evaluation

In this subsection, I conducted a subjective evaluation to confirm the validity of the theoretical analysis described in the previous section. In evaluation, noisy signals were generated by adding noise signal to target clean speech signals with a SNR of -5, 0, 5 and 10 dB. The noise signal was white Gaussian noise, babble noise, railway station noise, and factory noise, I presented by the proposed a pair of 10-dB-NRR signals processed by the proposed method and commonly used noise reduction methods, i.e., non-iterative spectral subtraction, Wiener filtering [6] and the minimum mean-square error (MMSE) short-time spectral amplitude (STSA) estimator [7], in random order to 10 examinees, who selected which signal they preferred from the viewpoint of total sound quality, e.g., less musical noise, less speech distortion, etc. The noise power spectral density is dynamically estimated by using minimum statistics method[8].

The results of the experiment are shown in Fig. 3. From this result, it is revealed that the signal processed by proposed musical-noise-free noise reduction is preferred to those commonly used noise reduction methods.

IV. Conclusion

In this letter, I introduced a theory for musical-noise-free noise reduction. Also, I derived the optimal parameters satisfying musical-noise-free condition to find the fixed point in the kurtosis.

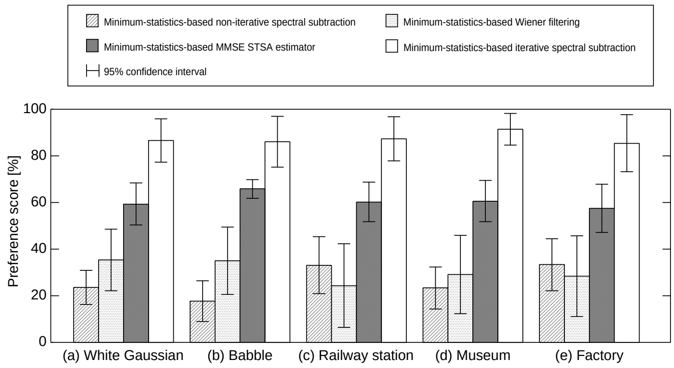

Fig. 3. Subjective evaluation results.

There exists researches using higher-order statistics to optimize sound quality, as well as described in the paper. In Refs. [9], [10], it is reported that such musical-noise-free condition holds even in Wiener filtering and MMSE-STSA estimator. Also, analyses for the method integrating microphone array and nonlinear noise reduction technique have been conducted [11], [12], [13].

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Feature Point Matching for Image Understanding and Scene Recognition

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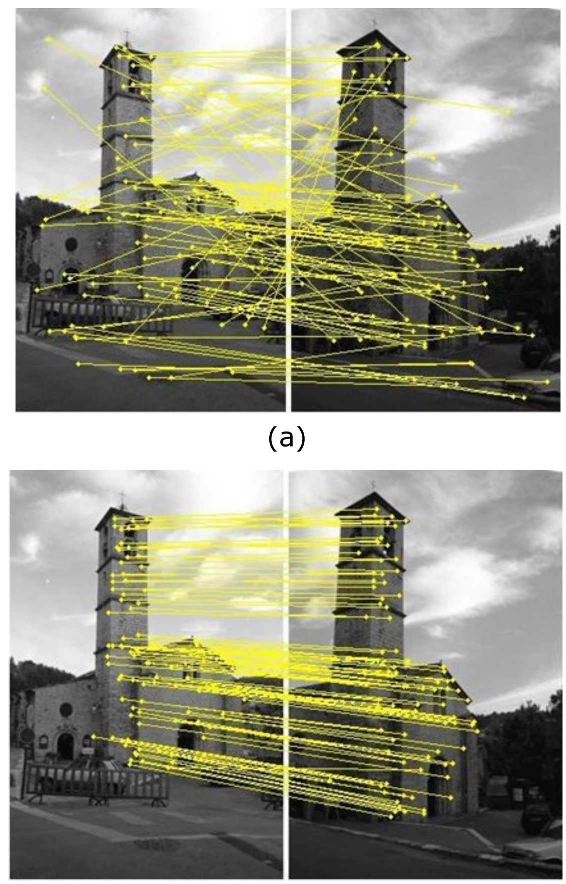
Image understanding and scene recognition has been an active research area in image processing and computer vision for decades. It continues to receive many technical challenges, since the resulted performance of system application oftentimes turns out to be far from satisfactory. Although the pixel-intensity values are the lowest-level data, it seems that many state-of-the-art image processing and computer vision applications need to begin with the so-called *feature descriptors* as their 'raw' data. Therefore, how to generate such feature descriptors to capture and represent the image contents as well as how to process these descriptors for various applications have aroused great interests to many researchers. In our laboratory, we have been investigating the latter issue recently and particularly focusing on *feature point matching*, which essentially aims to bestow machine with some capability or intelligence to automatically establish point-to-point correspondences across two images under matching. These images are normally acquired from the same scene but under different capturing conditions, such as viewing angles, scale changes, and so on. Such 'intermediate' level of processing becomes quite important to many image-based applications such as image registration, object detection and tracking, to name a few.

Despite various approaches that can be found in the open literatures, our approach is mainly based on graph theory. Indeed, our developed *graph-model-based* approaches have clearly delivered promising improvement in their respective applications. In our research endeavors, we have investigated two *graph-model-based* methods: 1) *bipartite graph* approach for identifying incorrect point-to-point correspondences from the already-established pairs [1], and 2) *directed graph* approach for performing common visual pattern discovery, which is basically establishing *object-to-object* correspondence pairs [2]. The common ground of these two works is that both are based on the so-called *feature descriptors* produced from the two images under processing. For that, the well-known SIFT feature descriptors [3] is adopted in our works, simply for the purpose of conducting simulations and demonstration, while other types of descriptors can be explored as well.

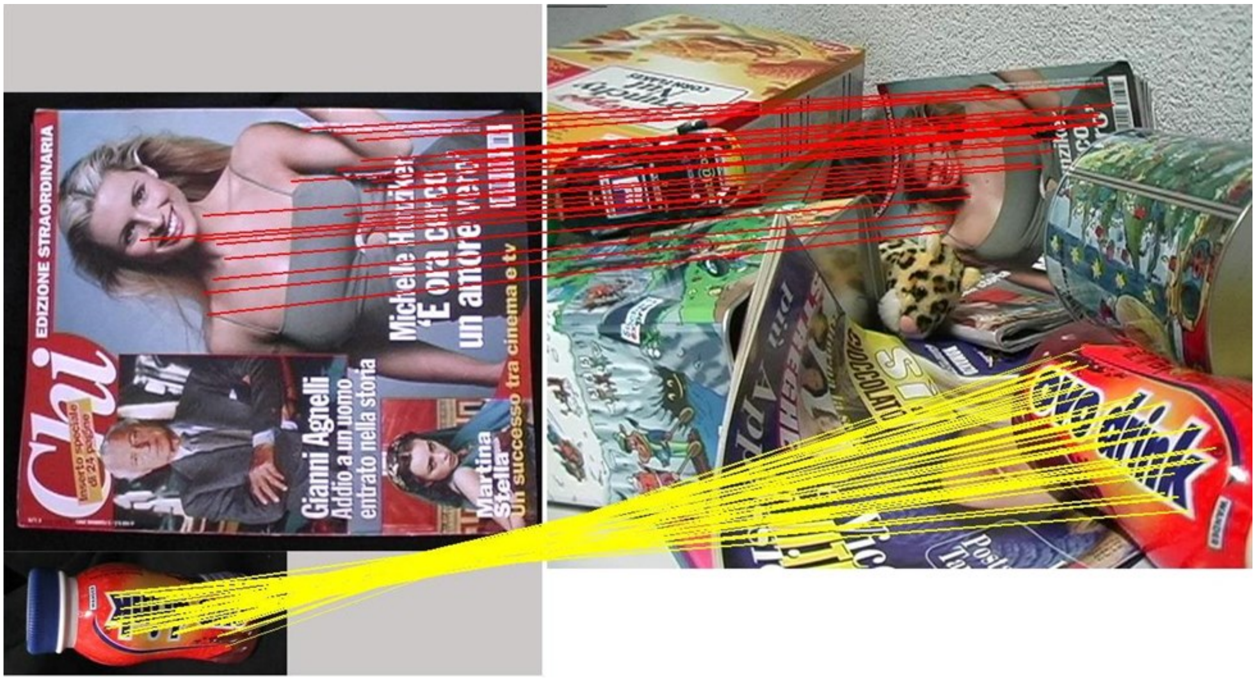
On the other hand, the main difference of these two works is that the first work is a post-processing technique, since it is used to detect *incorrect* establishments from the already established point-to-point correspondence pairs. Note that many applications tend to or even are required to generate as many feature-point



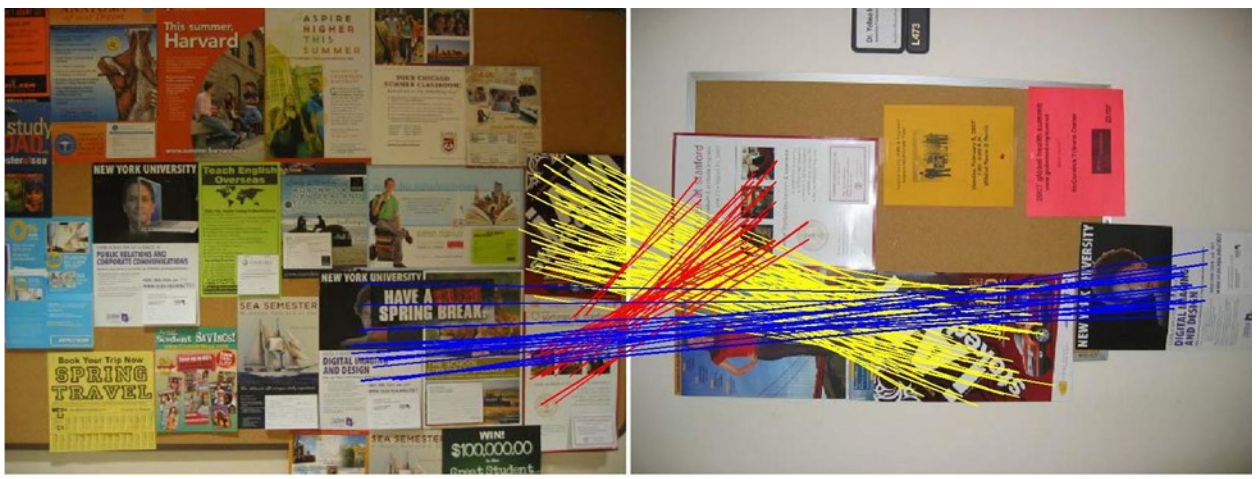
(a)

(b)

Figure 1: A demonstration of removals of many *incorrect* point-to-point correspondence pairs conducted from the wide-baseline image matching results. Note that only the top 100 point-to-point matching pairs are presented here by using: (a) the SIFT-based approach; (b) the proposed post-processing method. One can clearly see that many incorrect point-to-point correspondence pairs have been identified and removed, as the correspondence lines are fairly parallel among themselves in (b), compared with that in (a).



(a) Example 1



(b) Example 2

Figure 2: Two examples for demonstrating our developed *digraph*-based method on performing *common visual pattern discovery* task. Note that all commonly shared image objects have been successfully identified from the two images via point-to-point matching process, despite that these images were acquired under drastically different conditions, such as viewing angles, scales, and so on. The correctly identified objects are linked by a set of correctly established point-to-point correspondence pairs, as highlighted by different colors to denote various objects.

matching pairs as possible (e.g., image registration). As a result, many incorrect matching pairs will be inevitably generated. This might be due to the limitations of feature-descriptor generation algorithm and/or difficult image contents on representation. Therefore, a post-processing scheme to identify the erroneous establishments and remove them would become a highly welcomed 'tool' for conducting such remedy. By using our proposed mismatch removal method, many incorrect correspondence pairs can be identified and removed as illustrated in Figure 1.

Compared with the first work, the second work is completely opposite. First, it is *not* a post-processing approach; in fact, it tries to establish *correct* point-to-point matching pairs based on the feature descriptors from the very beginning. Second, the number of matching pairs could be established as least as possible. A representative application in this category is *common* *visual pattern discovery*, which basically performing *object-to-object* matching, rather than just *point-to-point* matching. For example, given two similar images as shown in Figure 2, how easy, or difficult for that matter, to you on identifying all common image objects simultaneously presented in both images? It is highly expected that this can even take for a while for the human being to conduct and with possible false establishments and/or miss-detections, let alone demanding machine to perform such task. To investigate this problem, the well-known SIFT descriptor representation has been the starting point to conduct feature-point correspondence task by using the *undirected* graph model approach. However, this often leads to many incorrect results (i.e., false detection and miss-detection), especially when the two images under matching have a large degree of viewpoint variation, severe cluttered background, and/or many repeated patterns (i.e., strong self-similarity). To improve the performance of common visual pattern discovery, a novel method by exploiting *directed graph* (or digraph) model has been successfully developed in our laboratory and with very promising results delivered as demonstrated in Figure 2 through two examples.

All the details and insights of the above-mentioned works can be found in [1] and [2]. In our view, many technical issues and novel image-based applications centering on the topic of feature descriptors are worthy to be further investigated.

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Myth or Fact: Frontier Journals Only Reject Bad Papers

Collated by Waleed Abdulla

I believe no one working in the research arena could claim immunity from being rejected when submitting publications to journals or conferences! It could be very stressful moment for the researcher when faced with such situation, especially with start-up researchers. The good news in this domain is to know that many prestigious and elite researchers and scientists have been rejected on weak grounds made by some editors and reviewers of prestigious publications. Editors in chief of highly regarded journals may sometime reject papers without even going to the review process either because they don't see enough interest in the papers or they may see them way too far from the common research practices. With no exaggeration to say some rejected papers by frontier journals stood later on for the Nobel Prize or very highly cited publications! For example, *Science* and *Nature* journals are widely considered to be the most prestigious multidisciplinary journals in the world, yet they did serious judgement mistakes about some Nobel Laureate papers.

In this short column I just indicate some of the examples to support the argument here. In this respect I will extract some of the examples from two papers depicted in the references.

The message that we want to convey here is that editors no matter how competent they are, they do mistakes and they are not always right. If the researcher believes in the conducted work, he/she may overrule the editors' decisions.

Examples of seminal rejected papers

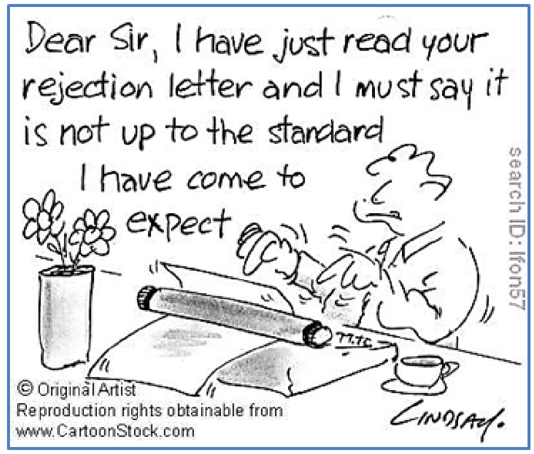
A paper by Russian physicist Pavel Cherenkov, reporting observations of previously unknown properties of visible radiation was rejected in the 1930s by Nature. The editors of Nature did not take the work seriously, but the editors of Physical Review accepted and published the paper in 1934. In 1958, P. Cherenkov,I. Frank, and I. Tamm shared the Nobel Prize in physics for the discovery and explanation of the Cherenkov effect.

William A. Fowler shared the 1983 Nobel Prize in physics. Fowler's best-known contribution to understanding the origin of the elements is a 1957 paper generally known as "BBF&H," after the names of the authors (Burbidge, Burbidge, Fowler, and Hoyle). According to the Swedish Academy of Sciences, the paper "is the basis of our knowledge in the field of nucleosynthesis". Nucleosynthesis is the process that creates new atomic nuclei from pre-existing nucleons, primarily protons and neutrons. This contributes to study the origin of the universe! However, the original version of the BBF&H paper was rejected by the first American journal to which it sent.

Both, Nature and Science rejected one of the first reports by Kary Mullis concerning the polymerase chain reaction (PCR), which turned out to become the most widespread method for analysing DNA. This was the discovery for which Mullis shared the 1993 Nobel Prize in Chemistry. Apparently, the editors of Science had not great faith in the revolutionary technique that was about to modernize the DNA analysis with its practical application to everyday life, and believed that the paper could be published in a secondary journal. As a consequence, the article appeared later in Methods in Enzymology.

Richard R. Ernst's work on nuclear magnetic resonance spectroscopy, which won the 1991 Nobel Prize in Chemistry, was rejected not once, but twice, in 1965 by the Journal of Chemical Physics. Afterward, it was accepted and published in the Review of Scientific Instruments! The editors claimed that the contents of originality were insufficient for publication in such journal. In consequence, Ernst had to publish his findings in the less known Review of Scientific Instruments. This article described the use of single, high energy pulses of radio waves containing all frequencies that would make atoms "flip" instead of a gradual sweep with a spectrum of radio waves that was in use previously. Also, Varian, a well-known fabricant of scientific instruments, resisted to build a spectrometer that incorporated the novel Fourier transform concept. As Ernst would later confirm, even they, the authors, did not foresee that the simple concept they were proposing could revolutionize Nuclear Magnetic Resonance.

In 1960, Theodore Maiman created the first working model of a laser, which used a ruby crystal pumped by a helical xenon discharge flash lamp. Maiman submitted a paper reporting the above findings to the *Physical Review Letters,* but it was rejected. A probable reason for this rejection was the editorial policy of rejecting papers that did not "contain significant contributions to basic Physics". *Nature* published an abbreviated form of his manuscript in 1960.



A paper authored by David S. Holmes and M. Quigley; reporting a rapid method for the preparation of plasmids, was initially rejected by Nucleic Acid Research. The discovery reported in this paper was done serendipitously (where fortunate discoveries made by accident) when Holmes heated a solution beyond the temperature at which the DNA contained in such solution would eventually denature. Surprisingly, the procedure worked and it was incorporated into a very popular compendium of recipes for molecular biology, receiving 2,325 citations from its publication in 1981 through 1993.

The Nobel Prize winning work of Paul Boyer has been rejected by Journal of Biological Chemistry. The work awarded with the 1997 Nobel Prize in Chemistry was the description of the molecular motor that creates cellular energy and the biochemical pump that transport such energy across membranes in cells.

Binning and Rohrer's report on their first experiments in scanning tunnelling microscopy, which earned them a Nobel Prize in Physics in 1986, was initially rejected on the grounds that it was "not interesting enough."

From looking at all these cases and there are way more than the mentioned cases, we can appreciate the bright side of the rejected papers. Yet, this is neither always the case nor it is the norm. The moral of the story, if researchers believe in what have been achieved then they should never give up after a rejection by one or more elite journals!

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Recent APSIPA Distinguished Lecture Activity

Lecturer: Professor Eliathamby Ambikairajah (UNSW Australia)

Title: Speaker Verification – The Present and Future of Voiceprint Based Security

Time: 3.00pm, 30th June 2014

Venue: Room 306B1, Faculty of Electrical & Electronic Engineering, Ho Chi Minh City University of Technology, Vietnam

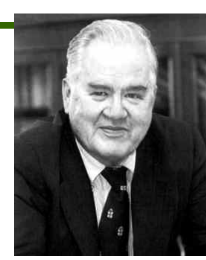
Host: Dr. Phu Le

Summary: Speaker verification refers to a system that analyses and understands an individual's voice, but more specifically their voice print, which can be used for security. Specifically, the use of voice prints to verify if the speech utterance belongs to the claimed speaker. This talk will provide an overview of how current text independent speaker verification systems are implemented as well as pointing out some emerging trends for the future.

Note: Lecture video available at: http://eemedia.ee.unsw.edu.au/contents/Ambi_APSIPA_DistinguishedLecture/


APSIPA Distinguished Lecture @ HCMCUT Vietnam

Quote of the Issue



Far better an approximate answer to the right question, which is often vague, than the exact answer to the wrong question, which can always be made precise.
John Tukey

John Tukey (1915-2000) was an American mathematician best known for development of the FFT algorithm.

PhD Thesis

Generation of a steerable audio beam from an ultrasonic transducer array

Chuang Shi

Department of Electrical and Electronic Engineering, Kansai University

Summary: This article is an abstract of the Ph. D. thesis of Dr. Chuang Shi, who obtained his Ph. D. degree from Nanyang Technological University, Singapore in May 2013 under the supervision of Dr. Woon-Seng Gan. The title of this thesis is "*Investigation of the Steerable Parametric Loudspeaker Based on Phased Array Techniques*". The full text of the thesis is available at this link or can be requested from the author.

Background

Sound is one of the most important means by which we communicate with our external environments. The demand for creating directional sound is growing in applications that necessitate personal privacy. For example, a directional sound system can employed to broadcast to a targeted zone of a library without disturbing the readers nearby. The directional sound system is also of significant importance in sound field control applications, including audio projection, spatial sound reproduction, immersive communication, and active noise control. In all the above applications, tailored equalization of the sound in both frequency and spatial domains is required.

Several techniques have been developed to implement the directional sound system, and each of them has its benefits and drawbacks. One of the ideas comes from the imitation of optical focusing lens. The world heritage site "Tian Tan" in Beijing, whose name means the altar of heaven, was built to enhance the sound at a specific zone. A smaller modern product adopting a similar architecture design is the acoustic imaging sound dome, which focuses sound waves to a confined listening zone. It is effortless to install but difficult to relocate the sweet spot after installation.

Alternatively, the loudspeaker array is another method to reproduce directional sounds. Although its radiation pattern is controlled based on the phased array techniques, the loudspeaker array needs a large size to obtain a sharp directivity at the low frequency band, which is typically lower than 2 kHz. Therefore, the loudspeaker array is not viable for a range of applications, where the installation size and the power consumption of the directional sound system are the crucial design criteria.

Hence, the directional sound system investigated in this thesis is known as the parametric loudspeaker. It is able to transmit a narrow sound beam from a compact ultrasonic emitter. An ultrasonic wave beyond human hearing range is utilized as a directional carrier. Thus, the resultant sound beam exhibits a similar beamwidth of the ultrasonic carrier. This sound principle is described by the parametric array effect in nonlinear acoustics.

If a steerable parametric loudspeaker can be realized, of which the radiation pattern is controllable, it will be more readily to be applied in the private messaging and a variety of applications of the sound field control. However, the radiation pattern of a parametric loudspeaker has not been satisfactorily described in theory yet. Thus, the directivity control methods of the parametric loudspeaker have not been investigated thoroughly. There are remaining aspects to be explored in this exciting research field. For this reason, this thesis focuses on the directivity control of the parametric loudspeaker based on phased array techniques.

Research Objective

The objective of this thesis lies in the following three aspects. Firstly, as a result of the sound principle of a parametric loudspeaker, its radiation pattern is not able to be controlled directly. Previous studies have merely assumed that the radiation pattern is a function of the directivities of the ultrasonic primary frequencies. So the steerable parametric loudspeaker, which adopts phased array techniques to achieve directivity control, is necessary to be investigated for its theoretical proof and simulation framework. In particular, the measurement setup requires careful examination to exclude errors introduced by the measurement system [1].

Secondly, the spatial aliasing problem of the steerable parametric loudspeaker is investigated [2]. Typically, the parametric loudspeaker transmits primary frequencies at 30 kHz to 50 kHz, of which the wavelengths are shorter than the diameter of the ultrasonic transducers. This restriction leads to the occurrences of spatial aliasing in the directivities of the primary frequencies. Solution to the spatial aliasing problem is obviously important to the design of the steerable parametric loudspeakers.

Thirdly, the radiation pattern of the steerable parametric loudspeaker based on phased array techniques is expected to be predictable. Two investigations are carried out in this thesis. The classic product directivity principle is modified to provide more accurate calculations of the radiation pattern of the steerable parametric loudspeaker based on the measured directivities of the primary frequencies [3]. Next, array errors in the steerable parametric loudspeaker are calibrated by a combination of the Monte Carlo method and the nonlinear least-squares method [4]. By this method, the simulated directivities of the primary frequencies result in closer matches to the measurement results, so that the radiation pattern of the steerable parametric loudspeaker is able to be further predicted with improved accuracy using the modified product directivity principle.

Contributions

This thesis focuses on the development of the steerable parametric loudspeaker based on phased array techniques. Its major contributions are highlighted in this section.

I. The feasibility of applying phased array techniques to carry out the steerable parametric loudspeaker is validated. The response of the steerable parametric loudspeaker is derived. Two beam steering structures are proposed. The validation of the product directivity principle is proven by a proposed transformation from the linear transducer array to the equivalent Gaussian source array. In particular, the measurement setup of the steerable parametric loudspeaker is presented with careful treatments of the microphone.

II. The grating lobe elimination effect is discovered in the steerable parametric loudspeaker. Even though grating lobes are commonly observed at the ultrasonic frequencies, they are rarely found at the resultant audible frequencies. This is an unique phenomenon of the steerable parametric loudspeaker, which is termed as grating lobe elimination. The grating lobe elimination is analyzed through both the simulation and the measurement results. The guidelines of designing the steerable parametric loudspeaker are derived, accounting for the occurrence of grating lobe elimination.

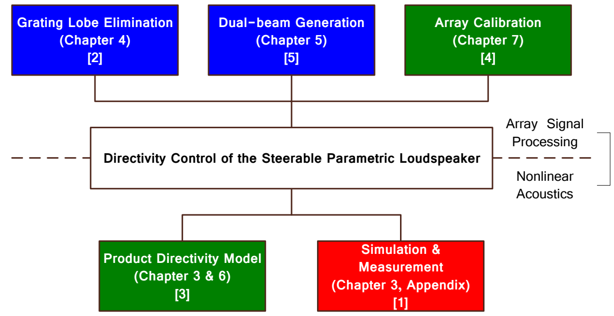


Fig. 1 The organization of Shi's thesis.

III. Dual sound beams can be generated from a single steerable parametric loudspeaker. A symmetric structure is proposed for the steerable parametric loudspeaker to generate two steerable sound beams without changing the hardware configuration. The second beam is created from the grating lobes of the primary frequencies. The angular separation between the two beams is simply adjusted by the frequency of the ultrasonic carrier. The grating lobe elimination is still effective in the symmetrical structure.

IV. Three modifications to the product directivity principle are proposed. The modified product directivity models are called as (i) the advanced product directivity, (ii) the exponential product directivity, and (iii) the combined product directivity, respectively. Configurations of the Gaussian source arrays that are equivalent to the directivities of the primary frequencies are incorporated into the product directivity models. It is shown by the measurement results that the proposed product directivity models are more accurate than the product directivity principle.

V. Array calibration method is developed for the steerable parametric loudspeaker. A structure including the system errors is proposed. The distorted radiation patterns with four types of system errors, namely the spacing error, the weight error, the delay error, and the steering angle error, are analyzed. A combination of the Monte Carlo and the nonlinear least-squares methods is also proposed to solve the calibration equation of the steerable parametric loudspeaker. The cross-validation results show significant improvement of the matching accuracies between the simulated and the measured radiation patterns of the steerable parametric loudspeaker.

Future Works

After the investigation of the steerable parametric loudspeaker reported in this thesis, there are several interesting extensions suggested to be explored in the future. For example, based on the product directivity models, the radiation pattern of the steerable parametric loudspeaker will be able to be designed with the inversed models. Thus, a directional microphone or an array of microphones can be deployed to monitor the sound field generated from the steerable parametric loudspeaker, and thus an adaptive feedback system can be proposed to steer the sound beam responding to the user or the environment. This adaptive system can be incorporated with ultrasound localization methods, such as the time reversal technique, to detect the targeted listener and implement a personal audio system with tracking capability.

Related Publications

[1] C. Shi and W. S. Gan, "Development of a parametric loudspeaker: A novel directional sound generation technology," *IEEE Potentials*, vol. 29, no. 6, pp. 20-24, November 2010.

[2] C. Shi and W. S. Gan, "Grating lobe elimination in steerable parametric loudspeaker," IEEE Transactions on Ultrasonics, Ferroelectrics, and Frequency Control, vol. 58, no. 2, pp. 437-450, February 2011.

[3] C. Shi and W. S. Gan, "Product directivity models for parametric loudspeakers," *Journal of the Acoustical Society of America*, vol. 131, no. 3, pp. 1938-1945, March 2012.

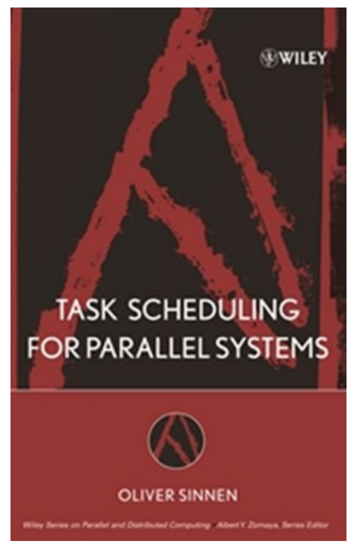
[4] C. Shi and W. S. Gan, "Analysis and calibration of system errors in steerable parametric loudspeakers," *Applied Acoustics*, vol. 73, no. 12, pp. 1263–1270, December 2012.

[5] C. Shi, E. L. Tan, and W. S. Gan, "Hybrid immersive three-dimensional sound reproduction system with steerable parametric loudspeakers," in *Proceedings of the 21st International Congress on Acoustics*, Montreal, Canada, June 2013.

Book Review

Task Scheduling For Parallel Systems

By: Dr. Oliver Sinnen, The University of Auckland



To execute a program consisting of several tasks on a parallel processor system, the tasks must be mapped to the processors and ordered for execution. This so called task scheduling is a very complex problem and crucially determines the efficiency of the parallel system. Parallel computing exists for many decades, yet programming a parallel system is still a challenging problem, much more challenging than programming a single processor system. With the proliferation of multi-core processors in many gadgets especially in mobile phones, the understanding of the foundations of parallel computing therefore become more important than ever. Thus, this book is devoted to task scheduling for parallel systems. It goes beyond the classic approach to task scheduling by studying scheduling under more advanced and accurate system models. These system models consider heterogeneity, contention for communication resources and the involvement of the processor in communication. This book is the first publication that discusses advanced system models for task scheduling in a comprehensive way.

For readers who may be new to task scheduling, the first chapters are essential. They serve as an excellent introduction to programming parallel systems, and they place task scheduling within the context of the program parallelization process. Then follow the basics of graph theory and the major graph models used to represent parallel programs. Next, the author introduces his task scheduling framework. He carefully explains the theoretical background of this framework and provides several examples to enable readers to fully understand how it greatly simplifies and, at the same time, enhances the ability to schedule.

The second half of the text examines both basic and advanced scheduling techniques, offering readers a thorough understanding of the principles underlying scheduling algorithms. The final two chapters address communication contention in scheduling and processor involvement in communications. Researchers, compiler and parallelization tool developers, and students will find this book very useful to them.

Reviewed by Waleed Abdulla

Asia-Pacific Signal and Information Processing Association Annual Summit and Conference 2014

December 9-12, 2014, Siem Reap, city of Angkor Wat, Cambodia

http://www.apsipa2014.org/

APSIPA ASC 2014

Call for Papers

Welcome to the APSIPA Annual Summit and Conference 2014 located in Siem Reap, city of Angkor Wat, the capital city in northwestern of Cambodia, and a popular resort town as the gateway to Angkor temples region. Siem Reap has colonial and Chinese-style architecture in the Old French Quarter. In the city, there are museums, traditional Apsara dance performances, silk farms, fishing villages and a bird sanctuary near the Tonle Sap Lake. The sixth annual conference is organized by Asia-Pacific Signal and Information Processing Association (APSIPA) aiming to promote research and education on signal processing, information technology and communications. The annual conference was previously held in Japan (2009), Singapore (2010), China (2011), USA (2012) and Taiwan (2013). The field of interest of APSIPA concerns all aspects of signals and information including processing, recognition, classification, communications, networking, computing, system design, security, implementation, and technology with applications to scientific, engineering, and social areas.

The regular technical program tracks and topics of interest include (but not limited to):

1 Biomedical Signal Processing and Systems (BioSiPS)

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- 1.3 Biologically-inspired Signal Processing
- 1.4 Medical Informatics and Healthcare Systems
- 1.5 Genomic and Proteomic Signal Processing

2 Signal Processing Systems: Design and Implementation (SPS)

- 2.1 Nanoelectronics and Gigascale Systems
- 2.2 VLSI Systems and Applications
- 2.3 Embedded Systems
- 2.4 Video Processing and Coding
- 2.5 Signal Processing Systems for Data Communication

3 Image, Video, and Multimedia (IVM)

- 3.1 Image/video Coding
- 3.2 3D image/video Processing
- 3.3 Image/video Segmentation and Recognition
- 3.4 Multimedia Indexing, Search and Retrieval
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- 4.1 Speech Processing: Analysis, Coding, Synthesis, Recognition and Understanding
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- 4.3 Audio Processing: Coding, Source Separation, Echo Cancellation, Noise Suppression
- 4.4 Music Processing

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- 5.7 Sparse Signal Processing
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- 5.9 Signal Processing for Energy Systems
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6 Wireless Communications and Networking (WCN)

- 6.1 Wireless Communications: Physical Layer
- 6.2 Wireless Communications and Networking: Ad-hoc and Sensor Networks, MAC, Wireless Routing and Cross-layer Design
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- 6.5 Devices and Hardware

Submission of Papers

Prospective authors are invited to submit either full papers, up to 10 pages in length, or short papers up to 4 pages in length, where full papers will be for the single-track oral presentation and short papers will be mostly for poster presentation. Conference content will be submitted for inclusion into IEEE *Xplore* as well as other Abstracting and Indexing (A&I) databases.

Important Dates

Submission of Proposals for Special Sessions, Forum, Panel & Tutorial Sessions	~~May 9, 2014~~	June 6, 2014
Submission of Full and Short Papers	~~June 6, 2014~~	July 4, 2014
Submission of Papers in Special Sessions	July 4, 2014	
Notification of Papers Acceptance	Aug. 29, 2014	
Submission of Camera Ready Papers	Sep. 26, 2014	
Author Registration Deadline	Sep. 26, 2014	
Tutorial Session Date	Dec. 9, 2014	
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Main Theme

Orange Technology, together with Green Technology, forms the double helical structure of sustainable human development. Instead of emphasizing the relations between environments and humans, as proposed by green technology, the orange technology aims to discover the relationship between human development and happiness, and bring more health, happiness, warming care, and more mental wellness to the society. We want to stimulate the establishment of orange technologies and to bring together scientists, engineers and other interdisciplinary professionals to share innovation ideas. The research scope includes computer science, electrical engineering, biomedical engineering, psychological/physiological science, cognitive science, social science, and dechnology (design + technology). The 1st IEEE International Conference on Orange Technologies (ICOT) was successfully held in Tainan, Taiwan in March 2013.

The 2nd ICOT will be held in Xi'an, one of the oldest cities in China, famed equally with Athens, Cairo, and Rome as one of the four major ancient civilization capitals in the world, on September 20-23, 2014. More than 3,100 years of history gives the city eternal charm. It is the eastern terminus of the Silk Road and home to the Terracotta Army of Emperor Qin Shi Huang. The ICOT2014 aims to continuously promote the related research, developments and applications of orange technology. Authors are encouraged to submit original papers on but not limited to the following topics:

Health Technology

- Biomedical Engineering and Applications
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- Friendly and Affordable Human-Machine Interface for Senior and Children Care
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- Affective Computing for Happiness Detection
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- Smiling Faces and Laughter Detection
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- System Design for Happiness Promotion
- Theory and Measurement of Gross National Happiness (GNH) Index
- Cultural Difference and Cross National Comparison in GNH
- National Policy Making and Strategies for Enhancing GNH

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Prospective authors are invited to submit full-length, four-page papers, including figures and references, in IEEE two-column format (www.ieee.org/conferences_events/conferences/publishing/templates.html). All ICOT papers will be handled and reviewed electronically. The ICOT 2014 website (www.saiip.org/icot2014) will provide you with further details. All the accepted papers will be indexed by EI and IEEE explore.

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Acceptance Notification of Special Session Proposals:	**March 25, 2014**
Paper Submission Deadline for Regular & Special Sessions:	**May 20, 2014**
Notification of Paper Acceptance:	**June 27, 2014**
Submission Deadline for Final Camera-ready Papers:	**July 5, 2014**
Deadline for Author Registration:	**July 27, 2014**

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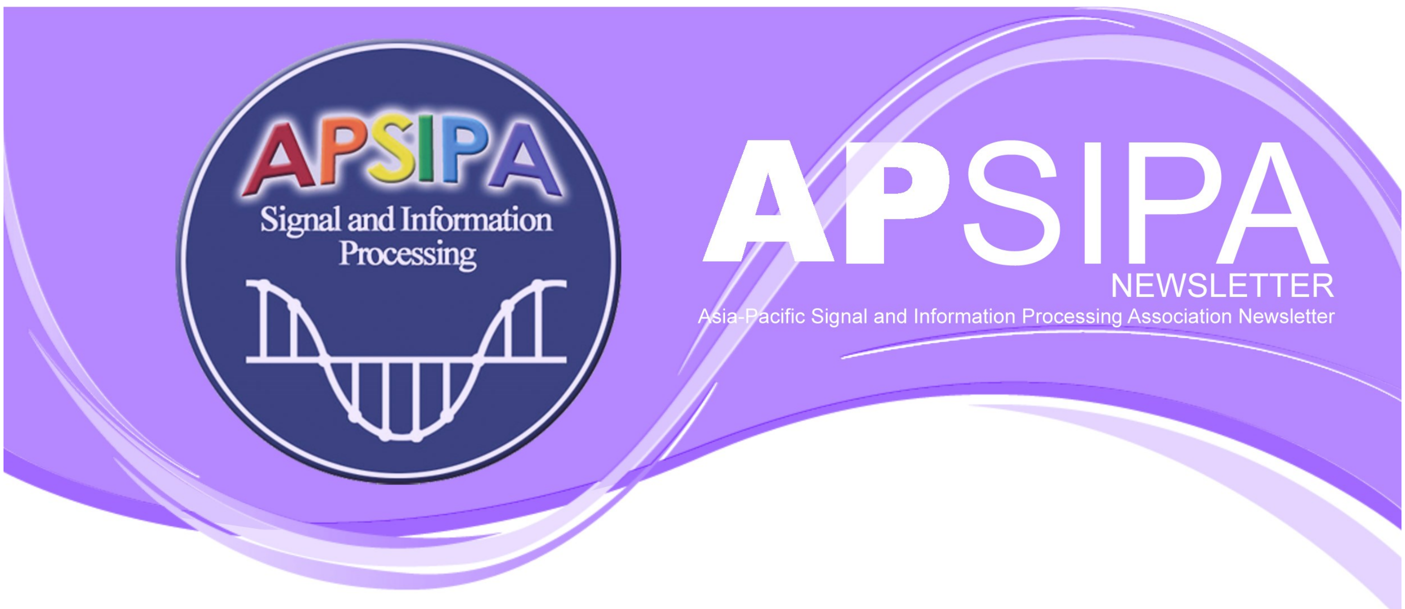

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