Issue 4

August 2013

APSIPA NEWSLETTER

Asia-Pacific Signal and Information Processing Association Newsletter

Distinguished Lecturer Program

This year, we will celebrate the second year of our APSIPA Distinguished Lecturer (DL) Program. I am so excited to give you an update of the DL activities as we gear up for APSIPA's another Annual Summit and Conference in Kaohsiung.

APSIPA established the DL Program in 2011 to serve its communities by organizing lectures given by distinguished experts. It is part of APSIPA's educational program that promotes the research and development of signal and information processing in Asia Pacific region. Particular attention is given to the specific needs of academia, and professionals in industry and government in developing countries. In 2012, we appointed 10 lecturers for the term of 2012-2013; In 2013, we appointed another 9 lecturers for the term of 2013-2014.

I would like to thank all the Distinguished Lecturers for their effort to promote APSIPA. We have received many encouraging feedbacks since the debut of the program. Our DLs have made a difference to the community. In 2012, 8 of them travelled to 9 countries/regions to deliver 29 lectures; In 2013, we are now 7 months into the year, 7 of them have travelled to 6 countries/regions to deliver 11 lectures. As of today, the 2012-2013 class of DLs have delivered 32 lectures, while the 2013-2014 class of DLs have delivered 8 lectures. Details can be found at APSIPA official website at [link].

APSIPA President awards a certificate of appreciation to those who have completed their DL duties during their terms at APSIPA ASC. During APSIPA ASC 2012, 5 DLs 2012-2013 (Mrityunjoy Chakraborty , Jen-Tzung Chien, Li Deng, Weisi Lin, Thomas Fang Zheng) received the certificates. Congratulations!

It is my great honour to serve as the first coordinator of APSIPA DL Program. As I pass the torch to Professor Thomas Zheng, the new DL coordinator 2013-2015, I would like to take this opportunity to express my gratitude for the support from our community. I have every confidence that the our DL Program will continue to flourish and bear fruits.

Finally, I look forward to meeting you again in APSIPA ASC 2013 in Kaohsiung, Taiwan in October!

Haizhou Li

APSIPA DL Coordinator 2011-2013

APSIPA ASC 2013

APSIPA ASC 2013 will be the fifth annual conference organized by Asia-Pacific Signal and Information Processing Association (APSIPA). Founded in 2009, APSIPA aims 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), and USA (2012). 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.

In this issue

Distinguished Lecturer Program 2013-2014

A. Masayuki Tanimoto

Emeritus Professor, Nagoya University

Senior Research Fellow, Nagoya Industrial Science Research Institute

Four DLs on FTV (Free-viewpoint Television) were given in Paris, Tokyo and Chengdu. The presentation slides are available at [link]

Lecture 1:

Date: February 22, 2013

Venue: The ARTEMIS (Advanced Research and TEchniques for Multidimensional Imaging Systems) Department, Telecom SudParis, Paris, France

Chair/Host person: Prof. Marius Preda

Lecture 2:

Date: June 12, 2013

Venue: Istituto Italiano di Cultura, Tokyo, Japan

Chair/Host person: Prof. Vito Cappellini

The DL was given at a teleconference between Tokyo and Florence, Italy.

Lecture 3:

Date: May 23, 2013

Venue: Automation School, University of Electronic Science and Technology China, Chengdu, China

Chair/Host person: Dr. Lu Yang

Lecture 4:

Date: May 24, 2013

Venue: Institute of Image Processing, University of Electronic Science and Technology China, Chengdu, China

Chair/Host person: Prof. King Ngi Ngan

Institute of Image Processing, University of Electronic Science and Technology China, Chengdu, China

B. Chung-Hsien Wu

National Cheng Kung University, Taiwan

Lecture 1:

Title: Why Emotional Awareness Matters: Recognizing the Power of Emotions from Facial Expression, Speech and Language

Date: March 12, 2013
Venue: 1st Lecture Room, Int. Conf. Hall, Kuang-Fu Campus, NCKU, Taiwan (Coincide with 2013 International Conference on Orange Technologies)

Chair/Host person: Prof. Jhing-Fa Wang

Professor Chung-Hsien Wu presenting his lecture

The audience asked the questions about how to collect and annotate the corpus. They also suggest the use of a commonly used corpus is beneficial for performance evaluation. Other suggestions are related to evaluation, which raised the problem on this research for how to apply the system to improve e-Learning. There are many discussions on this issue and the audience enjoyed the talk and discussion.

Lecture 2:

Title: Recognition of Emotion in Speech Using Structural and Temporal Information

Date: June 29, 2013
Venue: National Central University, Chungli, Taiwan

(Coincide with 2013 Speech Signal Processing Workshop)

Chair: Prof. Jia-Ching Wang

The audience asked the questions about if the speech content is useful for emotion recognition. They also commented that emotion will change with time. Many issues have been discussed and the audiences are satisfied with the lecture.

C. Tatsuya Kawahara

Kyoto University, Japan

Lecture 1:

Title: Recent trend of spoken dialogue systems

Date: March 4, 2013
Venue: Institute of Information Technology (IOIT), Vietnam Academy of Science and Technology (VAST), Hanoi, Vietnam
Chair: Prof. Luong Chi Mai
The talk gave a brief history of spoken dialogue systems including their evolution and recent technical trend. I pointed out there are two major approaches to dialogue systems: one is an AI (Artificial Intelligence) approach and the other is the statistical one. They are now converged as exemplified by Siri. But the current system is not intelligent enough to make a long interaction with users.
Lecture 2:

Title: Multi-modal sensing and analysis of poster conversations toward smart posterboard
Date: March 6, 2013
Venue: International Research Institute MICA, Hanoi University of Science and Technology (HUST)Hanoi, Vietnam
Chair: Prof. Tran Do Dat and Prof. Eric Castelli
This is a part of the joint symposium between MICA and School of Informatics, Kyoto University. We are developing a smart posterboard that can sense human behaviors via cameras and a microphone array and annotate key interaction events during poster sessions, which are held in many conferences including APSIPA ASC. Our focus is on the audience's feedback behaviors to investigate whether we can predict the interest and comprehension level of the audience. This is a very challenging and interesting project.

Professor Tatsuya Kawahara presenting his lecture

APSIPA Technical Committees

APSIPA has six technical committees that were established to promote and achieve the technical objectives of the Association.

- Signal Processing Systems: Design and Implementation (SPS)
  http://www.apsipa.org/TC/SPS.html
- Signal and Information Processing Theory and Methods(SIPTM)
  http://www.apsipa.org/TC/SIPTM.html
- Speech, Language, and Audio (SLA)
  http://www.apsipa.org/TC/SLA.html
- Biomedical Signal Processing and Systems (BioSiPS)
  http://www.apsipa.org/TC/BioSiPS.html
- Image, Video, and Multimedia (IVM)
  http://www.apsipa.org/TC/IVM.html
- Wireless Communications and Networking (WCN)
  http://www.apsipa.org/TC/WCN.html

Signal Processing Systems: Design and Implementation (SPS) TC

TC Chair: Takeshi Ikenaga (Dec/2014), Waseda University (Japan)

Signal Processing Systems: Design and Implementation Technical Committee (SPS TC) is promoting advancement and exchange of the research fields of design and implementation related to signal processing systems. Our fields of interest cover very wide areas, such as analog / digital, hardware / software and multimedia / communication systems. It also includes design methodologies and CAD tools for signal processing systems. We believe Asia-Pacific region plays a very important role from both academia and industry points of view. We welcome everyone to join!

Speech, Language, and Audio (SLA) TC

TC Chair: Chung-Hsien Wu (Dec/2014), National Cheng Kung Univ. (Taiwan)

As the new chair of the APSIPA Technical Committee on Speech, Language, and Audio (SLA), I would like to thank you for the opportunity to serve in this capacity. It is my great honor to provide this service to the SLA community. I would like to express my gratitude to Professor Lin-shan Lee, the past mentor of SLA TC, and Professor Tatsuya Kawahara, the past TC chair as well as to all contributors to our technical community. I would also like to introduce to you the new TC co-chair, Professor Changchun Bao, Beijing Univ. Technology, China, and new TC Secretary, Professor Hsin-Min Wang, Academia Sinica, Taiwan. The goals of our TC for 2013 include seeking new members in our community and encouraging young researchers to consider speech, language, and audio processing in their work. The nomination and election of the new TC members is also an import work in 2013. Currently, we have 35 TC members from 13 countries and very few members are female and from South/Southeast Asia, New Zealand, Canada, etc. Please encourage the researchers of these countries/areas to join our TC. I really appreciate your efforts to activate our community even more in the future. The other work we need to do is to promote APSIPA Annual Summit and Conference 2013, which will be held in Kaohsiung, Taiwan. The TC members are encouraged to help with the review of the submitted papers in SLA areas. In addition, we are planning to disseminate information about TC activities in our existing web site as well as to add content to the web site administered on our behalf by the APSIPA. In closing, I would like to express my gratitude for your support in my role as TC chair. I encourage you to contact me with any news or concerns you have regarding TC activities. I look forward to representing the TC and its interests in the APSIPA.

Biomedical Signal Processing and Systems (BioSiPS) TC

TC Chair: Yodchanan Wongsawat

(Dec/2014), *Mahidol Univ. (Thailand)*

According to the current world situation, the era of computer trends to move to the era of healthcare. All technologies and researches have the tendencies to be developed in order to enhance the human quality of life. The emerging technologies trend to be more healthcare-inspired.

Through the great success to get together the leading researchers in signal processing during the last 4 years by Dr. Toshihisa Tanaka, the past chair of BioSIPS, BioSIPS aims to produce the healthcare-inspired signal processing research on biomedical signal processing and systems including

1. Biomedical Signal and Information: Theory and Methods,
2. Medical Information and Telecare Systems,
3. Neural Systems and Applications,
4. Bio-inspired Signal Processing and Systems,
5. Biomedical Circuits and Systems.

As the 2nd BioSIPS technical committee (TC) chair, with the help from the vice chair, Dr.Tomasz Rutkowski, and all TC members, I am ready to move BioSIPS APSIPA to the leading biomedical signal processing and systems research community. With our strategy to enhance the international collaboration among our TC members through our internal BioSIPS workshop (supported by APSIPA) annually hosted by each of our TC members before the APSIPA conference, we hope to increase the number of paper submission, increase the number of APSIPA participants as well as increase the number of our TC members. The collaborations include the student exchange program, student competition, faculty exchange program, joint research program, and joint funding program. The goal of these collaborations is to let all participants know the benefits of being the BioSIPS. All BioSIPS activities will be promoted through the official BioSIPS APSIPA website. Finally, I am also most grateful to all BioSIPS participants that accept our invitation through the special sessions, tutorial session, as well as through the normal submission process of BioSIPS APSIPA. I assure that all participants will have the memorable experience with our research community.

Wireless Communications and Networking (WCN) TC

TC Chair: Tomoaki Ohtsuki

(Dec/2014), Keio University (Japan)

It is my great pleasure to be a chair of Wireless Communications and Networking (WCN) Technical Committee. It is also my great pleasure to work with Dr. Sumei Sun, a new WCN TC vice chair. Thanks to the great effort of former WCN TC chair, Prof. K.C. Chen, we had very successful events at past APSIPA Annual Summit and Conferences. We had a lot of papers and attendees. As you know, signal processing plays a major role in WCN. We need help of APSIPA members to enhance WCN supporting our life. We are looking forward to working with you.

APSIPA in Quick!

APSIPA Mission: To promote broad spectrum of research and education activities in signal and information processing in Asia Pacific.

APSIPA Conferences: ASPIPA Annual Summit and Conference.

APSIPA Publications: Transactions on Signal and Information Processing in partnership with Cambridge Journals since 2012; APSIPA Newsletters.

APSIPA Social Network: To link members together and to disseminate valuable information more effectively.

APSIPA Distinguished Lectures: An APSIPA educational initiative to reach out to the community.

An introduction to time-frequency clustering-based approaches to blind source separation

Ingrid Jafari, Roberto Togneri

School of EEC Engineering, The University of Western Australia

The human cognitive system displays a remarkable quality in its ability to distinguish between multiple, simultaneously active sources of sound. This capability has been extensively studied within the speech signal processing community, as automatic speech processing systems aim to perform at a comparable level. However, such systems are yet to perform at a level akin to human proficiency and are thus frequently faced with the cocktail party problem [1]. Blind source separation (BSS) algorithms aim to recover the original signals from just the mixtures with minimal information about the mixing process and acoustic environment. BSS is incorporated in many important disciplines ranging from audio signal processing to biomedical applications.

Blind source separation

Recently, the research held of BSS has evolved significantly to be an important technique in acoustic signal processing. The general BSS problem can be summarized as follows. M observations of N sources are related by the equation

$$X=AS \tag{1}$$

where X is a matrix representing the M observations of the N sources contained in the matrix S, and A is an unknown matrix that describes the mixing process. The aim of BSS is to recover the source matrix S given only the observations in X. The number of microphones relative to the number of sources present determines the classification of BSS: even determined ($M = N$), over determined ($M > N$) or underdetermined ($M < N$). The even determined system can oft be solved by a linear transformation of the data, whilst the over determined case can be solved by an estimation of the mixing matrix A. However, due to its noninvertible nature, the underdetermined BSS problem cannot be resolved via a simple mixing matrix estimation, and the recovery of the original sources from the mixtures is considerably more complex than the other aforementioned BSS classifications. As a result of its intricacy, the underdetermined BSS problem is of growing interest in the speech processing field.

Standard approaches to BSS

Standard techniques in BSS include those based on independent component analysis (ICA) [2] and the frequency domain clustering based techniques [3-5]. ICA-based approaches to BSS aim to find a linear representation of the sources in the observation mixtures, and assume the constituent source signals are statistically independent. ICA depends on matrix inversion, and as thus the number of microphones must equate or exceed the number of speakers. This imposes a restraint on its applicability in many practical applications of BSS where there may be an underdetermined setting. Furthermore, whilst statistical assumptions hold well for instantaneous mixtures of signals, in most audio applications the expectation of instantaneous mixing conditions is largely impractical as the convolutive mixing model is more realistic.

Clustering-based approach to BSS

On the other hand, the clustering-based approach to BSS is of significance due to its applicability in all BSS scenarios, including underdetermined. The clustering-based approach to BSS relies on a time-frequency (TF) clustering and an assumed underlying sparseness of the constituent source signals, as initiated in [6]. The clustering is used in the generation of TF masks, which are applied to the mixtures in the TF domain to obtain an estimate of the original sources. The pioneering TF masking work in [6] consequently motivated a plethora of clustering-based separation techniques [7-11]. Of particular mention is the multiple sensors DUET (MENUET) algorithm [3], which extended the DUET to encompass an arbitrary number and arrangement of microphones (in contrast to the stereo DUET), and also automated the TF mask estimation through the use of the k-means clustering algorithm.

Advances in clustering-based BSS beyond MENUET involve additional stages/complexities; of particular mention is the approach by Sawada et. al in [4] which resulted in superior BSS performance. The algorithm employed a two-stage mask estimation approach: a frequency bin-wise clustering followed by a grouping of the separated frequency bin components classified as originating from the same source. The bin-wise nature of the clustering promotes robustness against room reverberation and immunity against spatial aliasing; however, it also introduces additional complexity in its inherent requirement for permutation alignment. Therefore, the MENUET has the advantage over the stage-of-the-art system in [4] in that the full-band clustering eliminates any requirement for an additional stage of alignment.

Limitations

Despite the successes of full-band clustering techniques such as MENUET, it is not without its limitations: most significantly, the k-means clustering is not very robust in the presence of outliers or interference in the data. This often leads to non-optimal localization and partitioning results, particularly for reverberant mixtures. Furthermore, binary TF masks, as employed in the MENUET, have been shown to impede on the separation quality with respect to musical noise distortions. In subsequent research, the authors of the MENUET suggested that fuzzy TF masking approaches bear the potential to reduce the musical noise at the output significantly [12]. In [13] the use of the fuzzy c-means (FCM) clustering for mask estimation was investigated in the TF masking framework of BSS. This approach integrated a fuzzy partitioning in the clustering in order to model the inherent ambiguity surrounding the membership of a TF cell to a cluster, where any ambiguity could be attributed to the effects of reverberation/noise. As such, an extension of the MENUET algorithm through the use of alternative clustering schemes could potentially improve the BSS performance. This was demonstrated in the works in [14], where the FCM demonstrated improved separation performance when used in lieu of the hard k-means for underdetermined BSS.

Alternative clustering schemes with MENUET

Figure 1 shows the basic outline of the scheme. The microphone observations are transformed into the STFT domain, after which suitable features are extracted (Araki et. al provide a comprehensive review of suitable features in [3]). These features are clustered using an appropriate clustering technique. The results of the partitioning by the clustering are used to form suitable TF masks. The masks are then applied to the observation mixtures to demix and recover the source signal estimates. The details of the algorithm are omitted for brevity; the interested reader is referred to [3, 14, 15].

Three different clustering techniques were compared for TF mask estimation in the context of the MENUET algorithm: the original k-means [16], Gaussian mixture model (GMM) based clustering and the FCM [17]. The inclusion of the GMM clustering is due to its documented use in similar BSS approaches [18-20], and also to provide another soft clustering approach for comparison against the FCM (the posterior probabilities were utilized as the TF masks).

Experimental evaluations

Figure 2 displays the room configuration used in evaluations. In a simulated enclosure, four English speaking sources were situated around a 3-element microphone array, where the interelemental spacing was 4 cm. The room reverberation was varied between 0 ms and 300 ms (see [14, 15] for details). Separation ability was measured with respect to the signal-to-interference ratio (SIR), a measure which provides an estimate of the relative mount of interference in the target source estimate. Figure 3 displays the results averaged over all evaluations. It is immediately evident that the two soft masking techniques, GMM and FCM clustering, improve the separation quality by a considerable amount. As the reverberation is increased, the merit of FCM is visible as it has a significant lead in performance. A smaller standard deviation is also noted for the FCM clustering, which suggests that the FCM delivers more consistent, and thus reliable, separation of the speakers. Therefore, we can deduce the FCM clustering is an improvement over the original k-means in the context of the MENUET algorithm for underdetermined BSS.

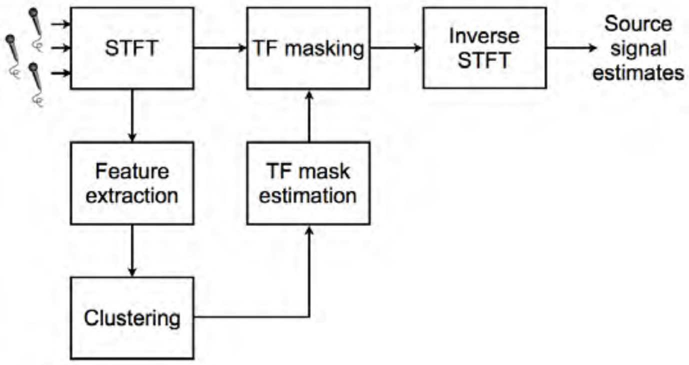

Figure 1: Generic flow of the TF clustering based MENUET scheme.

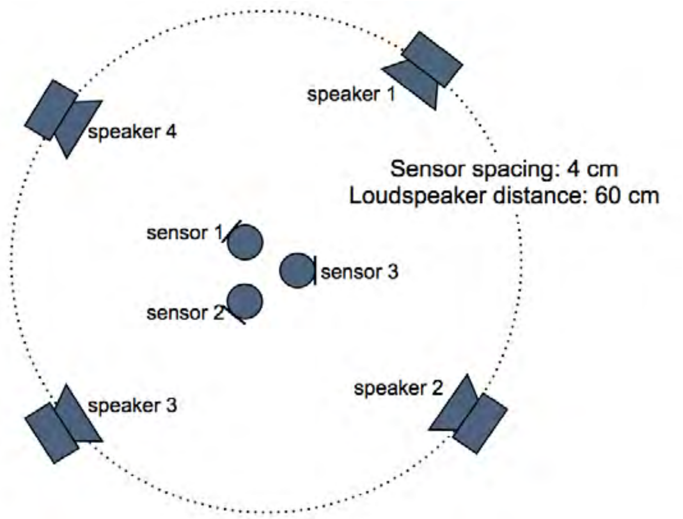

Figure 2: Room configuration for experimental evaluations.

Figure 3: BSS results for varying reverberation times. Source separation results compare three clustering techniques for mask estimation. Performance measured with respect to the SIR improvement (dB), where the average input SIR = -4.20 dB. The error bars denote the standard deviation.

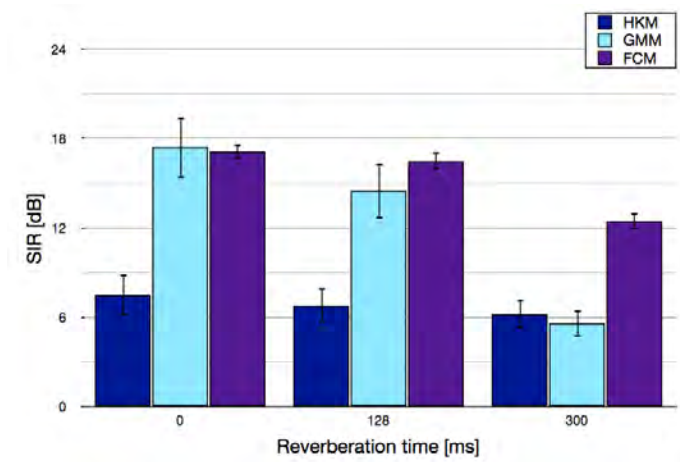

Figure 3: BSS results for varying reverberation times. Source separation results compare three clustering techniques for mask estimation. Performance measured with respect to the SIR improvement (dB), where the average input SIR = -4.20 dB. The error bars denote the standard deviation.

Future directions

Future research should focus upon the advancement of the TF mask estimation (i.e. clustering technique). Given the improvement when the k-means was replaced with

the FCM, a clustering paradigm with added sophistication and power may result in further improvements.

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Radio Wave based Sensing Using Array Sensor

Jihoon Hong and Tomoaki Ohtsuki

Graduate School of Science and Technology, Keio University,

3-14-1 Hiyoshi, Kohoku-ku, Yokohama, 223-8522, Japan

E-mail: hong@ohtsuki.ics.keio.ac.jp, ohtsuki@ics.keio.ac.jp

I. INTRODUCTION

In recent years, many countries have been facing the problem of a fast-aging population because of rising life expectancy and declining birth rates [1]. Cases where old people are staying alone are becoming very common. Remote monitoring in e-healthcare has become a hot issue in research fields and industries. To realize remote monitoring, the system should provide user's context including activities, locations, and environments' states, to doctors, nurses, and families. Currently, sensors have more attention for remote monitoring owing to low cost and high privacy protection, than video cameras [2]. However, one main drawback of the sensors such as accelerometer, gyro and pressure, is that the user must hold or be close to the sensors. It is inconvenient and gives stress. In this article we introduce our proposed sensing system using an antenna array that passively monitor user's context, referred to as array sensor.

II. WHAT IS ARRAY SENSOR?

Array sensor is an antenna array for monitoring the change of radio wave propagation of interest [4]. Compared to single antenna systems, the antenna array can distinguish signal and noise subspaces spanned by eigenvector. Because the signal eigenvector has robust to noise, it has more attractive feature than other signal features such as received signal strength (RSS) which may easily fluctuate over time by the noise. For example, the signal eigenvector changes over time only when the environment of interest changes, while RSS changes even when the environment does not change. The array sensor is shown in Fig. 1.

The most significant advantages of the array sensor are as follows. Array sensor can detect user's context over wide range even in non-line-of-sight (NLOS) It can be used without time and/or phase synchronization and calibration between transmitter and receiver. Moreover, it uses any frequency band on the application in use. Also, the array sensor does not require the accurate direction of arrival (DOA) information which requires complex signal processing algorithms, e.g., multiple signal classification (MUSIC). Last but not least, it is an effective system for user's context monitoring in privacy sensitive environments.

III. HOW DOES ARRAY SENSOR WORK?

The signal eigenvector can be extracted from correlation matrix of received array signal vector using the singular value decomposition (SVD) [3]. The array sensor uses a cost function that represents the correlation between the signal eigenvector obtained in advance as the reference vector and the signal eigenvector obtained in each time. Thus, the larger cost function represents the smaller change of the environment, while the smaller one represents the larger change. Similarly, we can also use the signal eigenvalue dependent on the application in use [5]. An example of cost function of eigenvector in different sampling rate is shown in Fig. 2.

Using the cost function of the signal eigenvector and/or ei-

Fig. 1: A photograph of array sensor

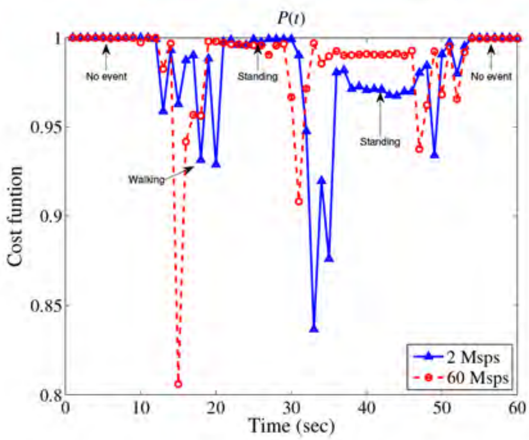


Fig. 2: An example of cost function of eigenvector

genvalue, the array sensor detects user's activities, locations, and environment's states based on a supervised learning such as a support vector machine (SVM). SVM has shown several advantages in prediction, regression, and estimation over some of the classical approaches in a wide range of applications owing to its improved generalization capabilities. Once the SVM has been trained, then all unknown samples can be classified in real time.

IV. EXPERIMENTAL RESULTS [6], [7]

A. Experimental Setup

The experiment was carried out in a complex area which is bounded by concrete walls, glass windows, and has a metal board within the deployment area as shown in Fig. 3. The transmitter needed for the array sensor was fixed in an NLOS position. The transmit antenna is a dipole antenna and uses a 2.4 GHz band. Received data from the array sensor are collected in a remote server via a wireless local area network (LAN). The experimental parameters of the array sensor are summarized as follows: sampling rate is 60 MHz, the number of snapshots is 1024, the reference received signal data for no event is taken before the experiments.

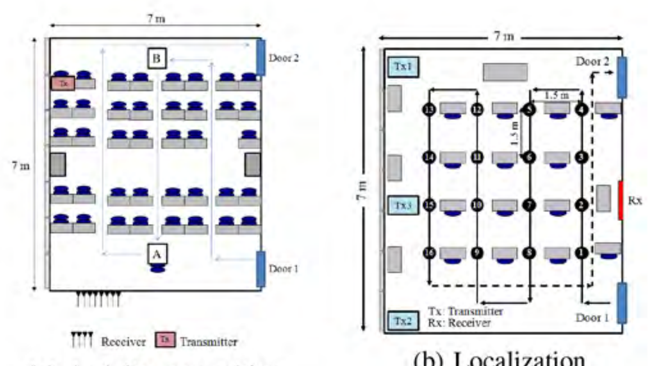
B. Activity Recognition

Table I shows the confusion matrix of the classification results from classification data. The numbers in this table show the number of classified data, and in particular the numbers in bold show the number of correctly classified data. We collected data with varying length of data for each state. Thus, the number of data for each state is different. We can see that the proposed method has high classification accuracy for some states: "no event", "walking", and "falling" states with 99.88 %, 94.54 %, and 73.06 %, respectively. The other states are classified over 50 % accuracy: "standing" and "sitting" with 66.42 % and 50.59 %, respectively. The average classification accuracy of these five states is 82.52 %.

C. Localization

In this experiment, we use three transmitters (Tx1, Tx2, and Tx3) with different frequencies to achieve more signal eigenvectors from multiple signal based on spatial smoothing processing (SSP). In total, 16 points are selected as candidate points shown as in Fig. 3. In the training phase, we obtained 100 observation data (approximately 15 seconds) when a person stands at each position. Five persons' training and testing data are used in the experiment. In the testing phase, a person enters the room from the door 1, walks from point 1 to 16 in the route indicated by solid arrows, stands at each position for 10 seconds, walks from position 16 to the door 2 in the route indicated by dot arrows, and exits from the room. The testing data can be obtained in real time and we localize the person in a continuous way with SVM.

Table II shows the root mean square error (RMSE) of the array sensor using different methods. We define the RMSE as the distance error between true position and estimated one. From these results in each method, we can see that the larger F shows the higher localization accuracy and the lower RMSE. This happens because SVM learning ability is improved by increasing the number of features. Thus, we can see that the proposed method that uses with SSP, shows better localization performance.



(a) Activity recognition (b) Localization

Fig. 3: Experimental environment

TABLE I: Confusion matrix of classification results

	Classified labels				
	No event	Walking	Standing	Sitting	Falling
No event	**846**	1	0	0	0
Walking	8	**1957**	56	23	26
Standing	2	60	**269**	20	54
Sitting	0	69	201	**302**	25
Falling	0	204	158	37	**1082**
Accuracy	99.88 %	94.54 %	66.42 %	50.59 %	73.06 %

TABLE II: Comparison of localization accuracy and RMSE. Ds = Dimension of the signal subspace, N = Number of sub arrays, F = Number of features

	Method	D_S	N	F	RMSE (m)
(a)	w/o SSP	1	0	2	2.71
(b)	w/o SSP	2	0	4	2.40
(c)	w/o SSP	3	0	6	1.92
(d)	w/ SSP	3	6	6	3.22
(e)	w/ SSP	4	5	8	2.11
(f)	w/ SSP	5	4	10	2.16
(g)	w/ SSP	6	3	12	2.06
(h)	w/ SSP	7	2	14	2.29
(i)	w/o SSP ∪ w/ SSP	3	6	12	1.67
(j)	w/o SSP ∪ w/ SSP	4	5	14	1.63
(k)	w/o SSP ∪ w/ SSP	5	4	16	1.66
(l)	w/o SSP ∪ w/ SSP	6	3	18	1.71
(m)	w/o SSP ∪ w/ SSP	7	2	20	1.47

V. CONCLUSION

In this article we introduced a novel sensing system using an array sensor. The array sensor is applicable for not only aforementioned applications but also many other applications including intruder detection.

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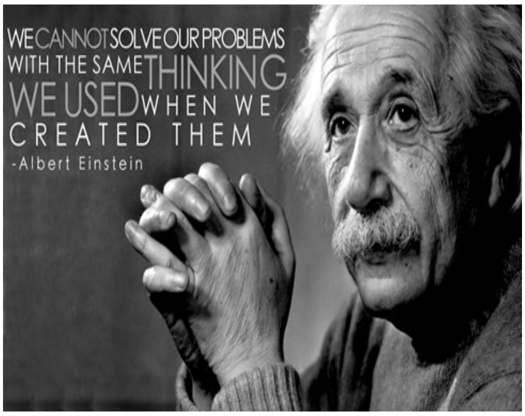
Albert Einstein Quotes



1. If we knew what we were doing, it wouldn't be called research.
2. In theory, theory and practice are the same. In practice, they are not.
3. Imagination is more important than knowledge.
4. One does not make wars less likely by formulating rules of warfare.
5. If the facts don't fit the theory, change the facts.
6. The hardest thing in the world to understand is the income tax.
7. I never worry about the future - it comes soon enough.
8. A person starts to live when he can live outside himself.
9. Anyone who has never made a mistake has never tried anything new.
10. Great spirits have often encountered violent opposition from weak minds.
11. Everything should be made as simple as possible, but not simpler.
12. Science is a wonderful thing if one does not have to earn one's living at it.
13. The secret to creativity is knowing how to hide your sources.
14. The only thing that interferes with my learning is my education.
15. Peace cannot be kept by force. It can only be achieved by understanding.
16. We can't solve problems by using the same kind of thinking we used when we created them.
17. Education is what remains after one has forgotten everything he learned in school.
18. The important thing is not to stop questioning. Curiosity has its own reason for existing.
19. With fame, I become more and more stupid - which of course is a very common phenomenon.
20. Two things are infinite: the universe and human stupidity, and I'm not sure about the universe.

Selected by Waleed H. Abdulla

APSIPA Transactions on Signal and Information Processing

The APSIPA Transactions on Signal and Information Processing is reaching this month the first anniversary of the publication of the first papers. Since then, papers are added to journal as soon as they are published and have been viewed thousands of times. The journal webpage provides detailed metrics of access for each of the papers:
http://journals.cambridge.org/action/displayJournal?jid=SIP

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Welcome to the APSIPA Annual Summit and Conference 2013 located in the Kaohsiung. It is the second largest city in Taiwan. The city's tourist attractions are located close to the harbor area. Just north of the harbor, Shoushan, or Monkey Mountain, has hiking trails that provide beautiful views of the city, and National Sun Yat-sen University is also inside the area. Kaohsiung always allows people to feel cozy and relaxed while bringing endless pleasant surprises at the same time. Kaohsiung residents, who are zealous and hospitable, like to keep their houses neat and tidy in order to warmly entertain their friends. When you enter Kaohsiung from the airport, station, highway, or harbor, you can always feel excited. What a sparkling city! Numerous modernized tall buildings, straight and clean roadways, and a green atmosphere seem to open a gate of grandeur and welcome visitors from everywhere.

APSIPA ASC 2013 will be the fifth annual conference organized by Asia-Pacific Signal and Information Processing Association (APSIPA). Founded in 2009, APSIPA aims 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), and USA (2012). 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. **Accepted papers in regular sessions and in special sessions will be published in APSIPA ASC 2013 proceedings which will be indexed by EI Compendex.**

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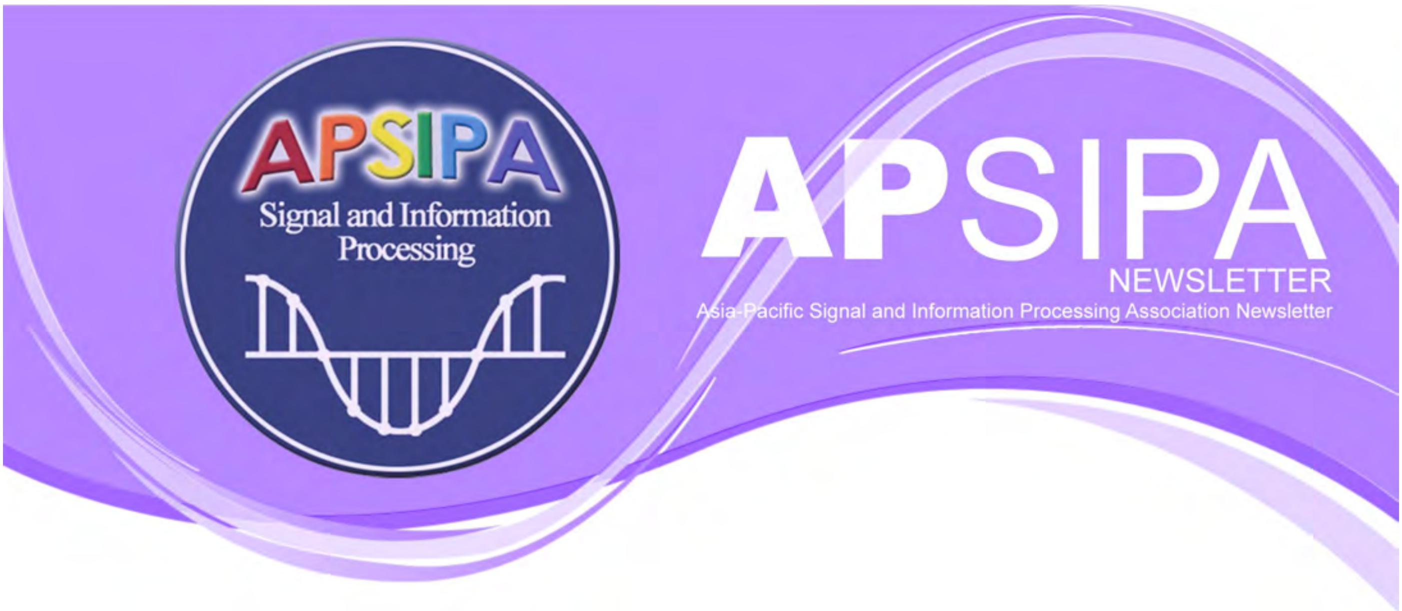

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