Poster and Demo Session

Bios and Abstracts

"Non-Orthogonal Multiple Access in Future 5G Networks with Correlated and Cooperative Sources"

Nazli Ahmad Khan Beigi,

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Bio: Nazli A. Khan Beigi received her B.Sc. degree from Sharif University of Technology, Tehran, Iran, in 2003, and the M.Sc. from the University of Science and Technology, Tehran, Iran, in 2005, both in electrical engineering. From 2005 to 2014, she worked as electrical engineer in industrial projects in Iran. She is currently working toward her Ph.D. degree in electrical and computer engineering at Concordia University, Montreal, QC, Canada. Her research interests lie in the area of wireless communications and include information theory, NOMA, source and channel coding, and their application in 5G systems.

Abstract: The huge tendency in employing 5G technologies including wireless sensor networks, Internet of Things (IoT) and mobile internet has brought about several challenges to manage, such as data traffic and massive connectivity in low latency, low power, and diverse service systems. In this regard, some solutions have been proposed, as in mm-wave channels, massive MIMO, ultra-dense networks, adaptive beamforming, and non-orthogonal multiple access (NOMA). In this paper, we investigate improving the cooperative NOMA in future 5G cellular networks by employing the correlation of user nodes' data in favor of increasing the channel's spectral efficiency. The considerable increase in spectral efficiency foreseen by our proposition is based on employing the limited communication channels in an efficient scheme based on information theory. The conventional wireless communication systems are based on allocation of orthogonal channels to different user nodes. This point-to-point direct communication discards any correlation between the sources. NOMA sets up a framework to share the resources of time, frequency, or codes in order to share the limited resources more efficiently. In communication schemes where user nodes have correlated sources, sharing the resources can bring about another advantage as well. The multiuser distributed source coding is the means to benefit from arbitrarily correlated sources by implementing higher data compression rates. In other words, due to advanced compression of data, more information can be transmitted per each bit of data, which we define it as higher throughput of data in this paper. Application of NOMA in multiuser systems has another achievement as well. Sharing the resources enables the user nodes to cooperatively transmit their data. This alternates the channel model to multiple access channels (MAC), instead of single user. Node cooperation in ad-hoc networks is already proved to result in higher throughput of data. In this paper, we use this scheme in the by-nature co-located multi-user 5G networks with massive number of user nodes as means of increasing the spectral efficiency. Our previous research works show how NOMA is a great capacity achieving method in multi-user networks. In this paper, we go a step further to consider the correlation and cooperation of user nodes as well. In a nutshell, we analyze the benefits of NOMA in 5G systems, with correlated sources, i.e. higher compression, and cooperation of sources, i.e. cooperative NOMA in a massive multiuser model, resulting in higher spectral efficiency.

"User Association and Load Balancing Model in an Ultra-Dense HetNet"

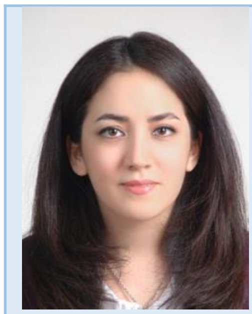
Dr. Edena Rakotomanana

Doctor in Electrical Engineering,
École de Technologie Supérieure

Bio: Edenalisoa Rakotomanana received the M.sc. in Electrical Engineering from Institut National de la Recherche Scientifique (INRS), Montréal, Canada in 2010. She received the Ph.D. in Electrical Engineering from École de Technologie Supérieure (ÉTS), Montréal, Canada in 2017. Her Ph.D. was part of the NSERC-Ultra Electronics Industrial Chair in Wireless Emergency and Tactical Communication. Her research interest covers resource allocation and load balancing in ultra-dense small-cell Heterogeneous Networks.

Abstract: The ultra-densification of the network and the massive MIMO system are considered as major factors of 5G since they promise huge capacity gains by exploiting the benefits of proximity, spectrum and spatial reuse. Both approaches are based on increasing the number of access elements per user, either by deploying more access nodes over an area or by increasing the number of antenna elements per access node. From the network point of view, the problem of choosing the best serving access node for each user, usually called user association, is very complex in ultra-dense networks (UDNs) because of the random topology deployment and the significant load variations observed in different access nodes. The problem becomes more significant when massive MIMO comes into play, since the presence of excess antenna elements per access node requires optimal exploitation of the available spatial degrees of freedom. In this work, our goal is to find an optimal user (UE) to antenna node (AN) balanced association in a small-cell dense heterogeneous network (HetNet) deployment that differs in terms of the number of transmit antennas and the number of served UEs in each block of resources. Specifically, we design new optimized techniques for exclusive UE-AN association in an UDN. In addition, we propose a fractional association of a UE to multiple ANs through new methods of ANs grouping and then a new selection technique of the best ANs group reaching the Pareto optimality via a multi-objective optimization. The proposed exclusive and fractional UE-ANs association techniques improve the load balancing in an UDN as well as increase the UE satisfaction in terms of allocated resources. As a result, there will be a significant increase in UE and overall network throughput compared to the methods proposed in the literature.

"Reducing the Decoding Complexity of the LDPC Codes"



Sareh Majidi Ivari,

Ph.D. Student,
Concordia University

Bio: Sareh Majidi Ivari received her M.A.Sc in Electrical and Computer engineering from Concordia University, Montreal, Canada in March 2017. During her masters, she worked on reduction of the decoding complexity of LDPC codes. She started her Ph.D. studies in Electrical and Computer engineering at Concordia University in May 2017. Her fields of interest are wireless communications, information theory, channel coding, source coding, data compression and video coding.

Abstract: Channel codes play an important role in the cellular communication systems. They are used to correct the communication errors and erasures that are caused by noise or interference. In 3G and 4G, Turbo code was used to provide channel coding. However, Turbo code offers high decoding complexity and the 3GPP standardization group is currently debating to replace the turbo code by Low Density Parity Check (LDPC) or polar code in 5G. The LDPC codes were introduced by Gallager in the early 1960's. Due to the good performance and simple and fast decoder, they received a lot of attentions. LDPC codes show good properties over the Binary Erasure Channel (BEC). Their performance is close to the Shannon capacity. They utilize iterative decoding algorithms. One important class of these algorithms is the belief propagation algorithm (BP). BP is a suboptimal decoder, but, approximates the maximum likelihood decoding. A significant research has been done to improve the performance of an existing LDPC code over the BEC. Guessing algorithm and Generalized Tree-structure Expected Propagation (GTEP) algorithm are some example of this approach. In the guessing algorithm, if the standard BP fails and cannot correct all the erasures, then the decoder makes some assumption on some of the erased bits. Check-sum determines if guesses are correct or not. In the guessing algorithm, the decoder complexity increases exponentially with increasing the number of guesses. GTEP has the complexity the same as BP. However, GTEP is not always successful. It depends on the position of the erased bits. In this paper, we present a new decoding algorithm. The new decoding algorithm is based on the guessing algorithm while decreases the complexity for a certain performance or increases the performance for a certain complexity. In the new decoding algorithm, instead of choosing a set of variable nodes and making assumption on them, the decoder chooses a set of check nodes and makes assumption on the variable nodes connected to the selected check nodes. This method decreases the number of possibilities, therefore, decreases the decoding complexity while improves the performance. Running the guessing algorithm to solve the remained erased bits right after the BP is not efficient. To decrease the required number of guesses in the guessing algorithm, the GTEP algorithm is run after the BP and before the guessing algorithm which is more sufficient. Since the complexity of the GTEP is less than the guessing algorithm, then it does not add a high decoding complexity. Therefore, the steps of decoding are: BP, GTEP, new decoding algorithm. At the end of two first steps, the probability of the remaining erased bits is reduced. If the decoder at each step cannot solve all the erased bits, the next step of decoding has to be run. The proposed decoding algorithm is applied to two regular half rate LDPC codes with lengths of ⟦10⟧^3 and 2×⟦10⟧^3. According to the density evolution, the theoretical threshold of the LDPC code with ensemble of (3,6) is 0.429. However, due to the cycles and finite code length, the actual rate is less than 0.429. With applying the new decoding algorithm, the performance of these two codes increases to 0.43 without increasing the complexity considerably. The number of possibilities in this paper is set to two.

"Beam Steering Components for 5G Applications"

Mohamed Ali,

PhD Student, Department of Electrical and Computer Engineering,
Concordia University

Bio: Mr. Mohamed Ali received the B.Sc. (with distinction) and M.Sc. degrees in electronics and communications engineering from Assiut University, Egypt, in 2010 and 2013, respectively. From 2010 to 2015, he was a Teaching and Research Assistant with the Department of Electronics and Communications Engineering, Assiut University. He is currently pursuing the Ph.D. degree in electrical and computer engineering from Concordia University, Montreal, Quebec, Canada, in 2016. He is a Teaching and Research Assistant with Concordia University. His current research interests include microwave reciprocal/nonreciprocal design and analysis and antenna design.

Abstract: With the improvement of mobile communication technologies and its broad applications, mobile communication will have more impact on the society. Such systems will support a variety of fundamental personal communication services as well as high-data-rate applications enabling high definition video transmission. These services would have the high potential for the industry and of support many sectors such as healthcare, connected homes, secure transport, security applications, and entertainment. To achieve such demands, many proposals associated with the development of 5G identify a set of requirements such as high data rate and larger coverage areas. This can be accomplished through utilizing the millimeter-wave frequency bands where more spectrums are available. Although high frequencies (mm-wave) can carry more data, they are limited in terms of their penetration capabilities and their coverage range. Moreover, such frequency bands limit deploying the traditional guiding technologies such as microstrip lines due to radiation or material losses. Therefore, utilizing the state of the art guiding structures, especially printed ridge gap waveguide (PRGW), is recommended. PRGW technology is considered among the essential guiding technologies due to the recently reported advantages. PRGW are featured with low signal distortion since PRGW supports a quasi-TEM mode. In addition, PRGW has low loss at millimeter-wave frequencies as the wave totally propagates inside an air gap. To deploy such promising technology, all types of mm-wave components must be developed especially directional couplers, which is a critical component in future systems. Directional couplers are a very important category of passive microwave circuits as they are mainly deployed to implement antenna beam-switching networks for power splitting and combining operations. This enables the space diversity, as well as high capacity. The main purpose of this work is to explore and develop PRGW directional couplers that can satisfy the requirements of today's advanced communication systems. In this work, the design of 3-dB planar quadrature hybrid couplers based on PRGW are presented. The proposed directional couplers are classified based on the coupling mechanisms as forward and backward couplers. They have an average size of 1.2〖 λ〗_g×1.3 λ_g with a relative bandwidth suitable for narrow (6%) and wide band (26.5%) applications. A systematic design procedure for the proposed couplers is presented and illustrated. These devices are featured with their low losses, compact size, and wide bandwidth. In addition, the fabrication of such designs is quite simple to provide cost-effective RF transceivers. The construction of the proposed couplers is based on a coupled PRGW lines connected by a central rectangular junction. The transition from the microstrip line to PRGW are essential parts to deploy the ridge gap waveguide technology in many communication systems. The proposed transition is designed based on tapered microstrip line matching section which used to validate the proposed coupler. The prototypes of the proposed couplers, as well as transition, are fabricated and tested, where the measured and simulated results show an excellent agreement. The proposed couplers have promising electrical characteristics that meet the future 5G required specifications.

"Model-based Correction of Ultrasound Image Deformations Due to Probe Pressure: Towards Experimental Validation"

Jawad Dahmani,

Ph.D. Student, Department of Electrical Engineering,
École de Technologie Supérieure

Bio: After an engineering degree in electronics from Polytech' Sophia Antipolis in France and a Masters degree in Mathematics applied to telecommunications, image and control signals, **Jawad Dahmani** led a research project in the field of signal processing for electric arc detection in electric vehicles. He is currently a Ph.D. student from the Electrical Engineering Department at École de technologie supérieure. His research interest covers 3D Freehand ultrasound and more precisely modelling the behaviour of soft tissues in order to compensate for image deformations arising from the variation of probe pressure on the skin.

Abstract: Freehand 3D ultrasound (US) consists in acquiring a US volume by moving a tracked conventional 2D probe over an area of interest. To maintain good acoustic coupling between the probe and the skin, the operator applies pressure on the skin with the probe. This pressure deforms the underlying tissues in a variable way across the excursion of the probe, which, in turn, leads to inconsistencies in the volume. To address this problem, we proposed a method to estimate the deformation field sustained by each image with respect to a reference deformation free image. The proposed method allows probe pressure compensation in US images using personalized estimates of the mechanical parameters of the tissues obtained from the images themselves, without knowledge of the spatial indentation caused by probe pressure and without a force sensor. The method is based on a 2D biomechanical model whose geometry consists of a rectangular background embedded with inclusions. The geometry of the different inclusions are segmented semi-automatically using an active contour method that is robust to US speckle noise. Once the geometry of the model is created from the reference image, boundary conditions are defined on this geometry. Rough initial estimates of the elasticity parameters used to drive the 2D model are drawn from the literature. The initial estimate of the indentation parameter used to drive the model is determined using a purely image-based estimate of the displacement field. The deformation field resulting from the model is used to compensate the deformed image. The optimal parameters are estimated by maximizing the mutual information between the reference and the corrected images using the Nelder-Mead simplex algorithm. Preliminary validation was conducted with synthetic US images generated using a 3D biomechanical model. Results show that the proposed method leads to more accurate deformation field estimates compared to a purely image-based method. Experimental validation of the method is currently in progress and is performed on phantoms with controlled mechanical (Young's modulus) and acoustic properties (velocity, acoustic impedance, attenuation and echogenicity). For this purpose, an easy and repeatable fabrication procedure to make mechanically tuned tissue-mimicking phantom is developed. A house-hold gelatin is used as a basis for the phantom, and a psyllium hydrophilic mucilloid fiber are added to get the echogenicity of the phantom. The mechanical properties of the phantom can be tuned by varying the concentration of the gelatin and refrigeration time of the sample. A wide range of elastic properties can be reached (6kPa-72kPa) in this way, allowing simulation of normal and abnormal human soft tissues with acoustic properties close to those of human organs. Controlled phantom deformation trials are performed using a robot allowing a vertical support of the US probe, and an accurate sub-millimeter displacement. A force sensor is also incorporated into a closed system with the probe, allowing the measurement of the force exerted by the probe on the phantom. The purpose of this experiment is, first, to validate our estimates of the elasticity and indentation parameters used to drive the biomechanical model, and second, to test our pressure compensation method by comparing the corrected image with pressure-free phantom images.

"Real-time Recognition of Suicidal Attempts and Drug Overdose Behaviour Using an RGB-D Camera"

Dr. Rafik Gouiaa,

Postdoctoral Fellow, Department of Electrical Engineering,
École de Technologie Supérieure

Bio: Rafik Gouiaa received a Ph.D. degree from the University of Montreal in 2017. His research interests include computer vision, video-surveillance and machine learning. He has been working on human action recognition from invisible shadows, fall detection and suicide by hanging detection. He is currently a postdoctoral fellowship at the École de Technologie Supérieure (ÉTS). He is a member of IEEE and the Treasurer of the IEEE Consumer Electronic Chapter (Montreal section).

Abstract: Inmates in solitary confinement attempt to harm themselves in many ways, resulting in trivial to mortal injuries. In this context, suicide by hanging and drug overdose are among the leading causes of death within the incarcerated. Rapid detection of hanging or drug overdose attempts can reduce the mortality rate and raise the chance to survive the event. In the last two decades, several technologies have been presented to detect and prevent the suicide attempts by hanging, but most of them use cumbersome devices, or they are greatly depending on human attention and intervention, which limit their applications. In this abstract, we propose a Kinect-based intelligent system to automatically detect the suicide by hanging attempts or drug overdose behaviour such as fall, abnormal gait, aggressive behavior etc. This system can be a good solution to prevent several tragedies in prisons.

"Emotion-Based Intelligent Guidance Planning for Aerial Robots"

Amin Haeri,

Ph.D. Candidate,
Concordia University

Bio: Amin Haeri is a PhD candidate at Concordia University. He is currently working in the areas of robotics and computational fluid dynamics with a focus of robot-soil interaction for planetary rovers under the supervision of Dr. Krzysztof Skonieczny. He received a MSc. at Sharif University of Technology in flight dynamics and control engineering; and a BSc. at Amirkabir University of Technology in aerospace engineering.

Abstract: In this research, an emotion-based intelligent guidance planning for flying vehicles is proposed. It utilizes different sciences including machine learning, neural, fuzzy and psychology accompanied by flight dynamics and modeling. The proposed system enables the robot to make emotion, learn and make decision based on its given personality
. The decision making is related to the path planning and how moving from the current position to the designed one. In other words, the robot, at each moment, knows what to do and how to do. Therefore, guidance commands of the flying robot are as outputs of the system. In the present research, the control system is assumed ideal. The introduced intelligent guidance system is divided into two phases: offline and online. First, the robot is trained for a specific mission by the learning subsystem offline. Then, the both emotion modeling subsystem and decision-making subsystem are added to the main system in the online phase. In this phase, the robot has the ability to choose optimal action and reach to the position by utilizing action-to-commands subsystem. In order to design and implement the system, a new mission is defined: how to train an aerial robot to surreptitiously reach the target; i.e. the robot should learn how to use static obstacles to be hidden from dynamic obstacles and touch the target.

"A Neural Network Chatbot for Farsi"

Farhood Farahnak,

Ph.D. Student, Computer Science at the CLaC Lab,
Concordia University

Bio: Farhood Farahnak obtained both B.Sc. and M.Sc. degrees in Computer Science from Shahid Beheshti University, Tehran, Iran, in 2013 and 2016, respectively. His master thesis was about developing a conversational model based on the deep neural. Currently, he is a Ph.D. student in Computer Science at the CLaC Lab, Concordia University, Montreal, QC, Canada. His research interests are Machine Learning, Artificial Intelligence and Natural Language Understanding and Generation.

Abstract: The field of Conversational Modeling (or chatbot) has experienced a significant interest in recent years both in the research community and in the industry, where they are used in many commercial applications ranging from technical supports to entertainment. However, most work in this area still use hand-crafted rules and are designed to handle conversations in a specific language and domain such as travel booking. Unfortunately, these rules are not easily portable to other languages or domain, hence significant effort is required to build a new set of rules for a new application. On the other hand, recent breakthroughs in Deep Learning for Natural Lan- guage Processing and the availability of datasets [1] has lead to the development of end-to-end systems that can overcome these issues of language and domain-specificity. The goal of our project was to develop an open-domain chatbot for Farsi. To do this, we trained a deep neural network with a sequence-to-sequence architecture. This type of network has shown great results in open-domain chatbots (eg. [3]) as well as in machine translation (eg. [2]). As training data, we used Persian subtitles of movies from www.opensubtitle.com. This dataset, after preprocessing and removal of non-conversational texts, contains almost six million sentences. To train our chatbot, we needed pairs of questions and responses, so we used every sentence in a subtitle as a response to the previous one. Using these pairs of question- response, we trained the sequence-to-sequence model to predict an appropriate response for a given question. The model consists of two main modules: an encoder and a decoder. The question is fed to the encoder and the output is used as the hidden state of the decoder. Then the decoder generates the response word-by-word. Using the measure of perplexity, our model for Farsi underperforms when compared to a similar approach for English [3]. Our model performs much better (in term of perplexity) for short responses but degrades with longer ones. We believe that a larger dataset and an objective function better adapted to the task may improve our result.

References

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[3] Oriol Vinyals and Quoc Le. arXiv:1506.05869, 2015.

"Language Model Utilization in a Persian P300 Speller"

Hessam Amini,

Ph.D. Student, Dept. of Computer Science, CLaC Laboratory, Concordia University

Bio: Hessam Amini received his B.Sc. in Computer Software Engineering from Ferdowsi University of Mashhad, Iran, in 2015. He received his M.Sc. degree in Health Information Technology from University of Tehran, Iran, in 2017. During his M.Sc. studies, under the supervision of Dr. Hadi Veisi, he worked on his research which was towards developing a Persian Brain-Computer-Interface-based (BCI) speller with language model utilization. He is currently a Ph.D. student of Computer Science at the CLaC laboratory at Concordia University, under the supervision of Dr. Leila Kosseim. His current research is focused on incorporating Machine Learning (ML) in Natural Language Processing (NLP) applications.

Abstract: A Brain-Computer Interface (BCI) is a device in which control commands are made only through brain signals. Spelling is an important application of BCI, as it allows users with speech or motor disabilities to communicate by typing their desired words using no muscle movements but only through cognitive tasks. Most of the state of the art BCI systems (e.g. EPOC, IntendiX, etc.) use non-invasive brain signal acquisition methods, for example ElectroEncephaloGraphy (EEG), in order to provide a safer, cheaper, and more comfortable interface for the user. However, these non-invasive methods suffer from low Signal-to-Noise Ratio (SNR), which significantly degrades the performance of the BCI system. In order to overcome this drawback, contextual information is often used as a post-processing step to complement the brain signal received. The goal of our project is to evaluate the use of language model as contextual information to improve a BCI speller's performance. Our BCI speller utilizes the P300 signal, which is a brain signal component that is elicited around 300ms after a subject perceives an unexpected external stimulus. In our project, we first designed a stimuli screen in order to present the P300 stimuli. Using the designed screen and the target, words which had previously been chosen, we recorded EEG data from 5 users. After preprocessing the data (including band-pass filtering, artifact removal using ICA, and creating vector samples), we classified the EEG samples as P300 and non-P300, using a Support Vector Machine (SVM) with a Radial Basis Function (RBF) kernel. We then computed the distances of samples from the decision boundary of our classifier. In the post-processing phase, the classification results and the distances from the the decision boundary of the classifier (both solely and in combination with a language model) were used to detect the target characters. We experimented with character-level n-gram language models with n = {1, 2, 3, 4, 5}. Results show that, as the value of n increases, the character detection accuracy of the BCI speller improves. Overall, without the language model, the average accuracy of the system reached 91.3%, but when complemented with a 5-gram model, the accuracy reached 93.2%. The use of the language model also helped the subjects reach 100% accuracy faster: without using the language model, the optimal accuracy for each subject was achieved within an average of 7.6 trials, while this number decreased to 5.6 trials when using the 5-gram language model.
Keywords: Brain-Computer Interface, P300, Language Models, n-gram Models

"Using N-gram Language Models to Identify Author's Mother Tongue"

Elham Mohammadi,

Master's Student, Dept. of Computer Science, CLaC Laboratory, Concordia University

Bio: Elham Mohammadi is a master's student of computer science at the Computational Linguistics at Concordia (CLaC) Laboratory, Concordia University, Montreal, under the supervision of Dr. Leila Kosseim. Having a computational linguistics background from her previous studies at University of Tehran, Iran, Elham's research is focused on the use of machine learning in natural language processing applications.

Abstract: Native Language Identification (NLI) is the task of determining an author's native language, based on a sample of his/her writing in a second language. In recent years, NLI has received much attention due to its challenging nature and its applications in language pedagogy and forensic linguistics (Kochmar, 2011). In language teaching, NLI can help in determining the role of native language transfer in second language acquisition, so that course designers can adapt the teaching material based on the native language of the learners (Laufer and Girsai, 2008). In forensic linguistics, NLI can be used as an important feature to identify the author of a text which is of great interest to intelligence agencies (Tsvetkov et al., 2013). The goal of our work is to evaluate the use of language models for the task of NLI. To do so, we used a mixture of character and word n-gram models trained on the dataset provided by the NLI Shared Task 2017 (Malmasi et al., 2017). The dataset consists of test takers' written essays and the transcriptions of their spoken responses to a standardized test of English proficiency. The dataset contains 13,200 texts (11,000 for training, 1,100 for evaluation, and 1,100 for testing), belonging to 11 different native languages (Chinese, Japanese, Korean, Hindi, Telugu, French, Italian, Spanish, German, Arabic, and Turkish). To identify the native language of the essay portion of the dataset, we experimented with a combination of 4-grams character-level, and unigrams and bigrams word-level probabilities. For the transcription of spoken responses portion of the dataset, we relied more heavily on lower level character n-grams and used a combination of trigrams and 4-grams character-level probabilities, and unigrams and bigrams word-level probabilities. We then classified the language with the highest final probability as the native language of the test taker. At the NLI Shared Task 2017 (Malmasi et al., 2017), we achieved an overall F1-score of 0.7748, using both he essay and the speech transcription datasets. The results show that, despite being simple, the n-gram language model is useful in the task of NLI. However, a deeper error analysis shows that some close pairs of languages are harder to discriminate, for example Hindi and Telugu, and more data would be required to train our model.

Keywords: Native Language Identification, Language Models, n-gram Models

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"Fully-automated tongue detection in ultrasound images"

Elham Karimi,

Master's Student,
École de Technologie Supérieure

Bio: Elham Karimi is a Master's student in Software Engineering working under the supervision of Professor Catherine Laporte. She obtained her Bachelor's degree in Computer Engineering from the University of Kurdistan in 2016. Her research interests are mainly focused on medical image processing and computer vision.

Abstract: Ultrasound imaging is used in speech studies to observe tongue motion. It is a very attractive measurement system because it is non-invasive and easy to use. Tracking the tongue in US images provides information about the shape and kinematics of the tongue. Most methods for segmenting/tracking the tongue require manual initialization or training a substantial amount of labeled images. We have invented a new method for segmenting a single tongue image which requires neither training nor manual intervention. In this presentation, we will show how this can be used to initialize more accurate tracking methods. Despite the fact the sound echoes reflected off the tongue usually account for brighter intensities in US images, detecting tongue related regions has many known challenges such as high speckle noise, acoustic shadowing, etc. To enhance US images in a way that they can emphasize the regions of tongue contours, we apply a phase symmetry filter at each frame of a given video sequence. Phase symmetry is a dimensionless quantity that is invariant to changes in image brightness or contrast and can be used as a ridge detector. Since it works better than gradient-based methods, we used this filter to expose the tongue in US images. As we are interested in bright regions that include tongue contour points, we set our next step to produce a binary image of the filtered frame from the previous step. This is done through an adaptive thresholding procedure and we try to identify those white regions that either include or are close to the tongue contour points; we call them regions of interest (ROI) in this work. To find ROIs, we generate several quadratic curves that could potentially look like tongue contours. As we are trying to capture mid-sagittal tongue contours, we sample three data points from three different regions in the image to fit each of these curves: around the bottom left, the center, and the bottom right of the frame. We intersect these curves with white regions and we find the curve with the most intersections. White regions that are not close enough to this curve are removed, and the others are assumed to be ROIs. The next step is to find the medial axis of the ROIs, which provides a set of candidate tongue contour points. The set of candidate points can ultimately be used to initialize a more accurate segmentation without manual intervention. When segmentation is performed offline, initialization does not need to take place in the first frame of the sequence. Therefore, we also developed two quality measures that are predictive of the reliability of our method so that we can choose an initial frame where we are confident that segmentation will work. These two scores include the average distance between pairs of consecutive points and the ratio of the coverage length of the candidate points on the x-axis over the frame width. In this work, we use the multi-hypothesis approach of Laporte and Ménard for tracking. We tried this approach on 346 video sequences with an average length of 51 frames, captured from 7 subjects. Our experiments show 87% correct initialization of tongue contour points which is a very promising result. As for future work, we hope to implement a real-time system that works online during speech therapy sessions.

"Image Denoising Using Adaptive Total Variation and Tchebichef Moment as Sparse Regularizer"

Dr. Ahlad Kumar,

Postdoctoral Fellow,
Concordia University

Bio: Dr. Ahlad Kumar is currently doing research in Concordia University, Montreal Canada in the field of image processing. His area of interest is in the field of image restoration. He received his PhD degree from University of Malaya, Malaysia in 2016. He is gold medalist during his M.Tech from ABV-IIITM in 2007.

Abstract: Structural information, in particular, the edges present in an image are the most important part that get noticed by human eyes. Therefore, it is important to denoise this information effectively for better visualization. Recently, research work has been carried out to characterize the structural information into plain and edge patches and denoise them separately. However, the information about the geometrical orientation of the edges are not considered leading to sub-optimal denoising results. This has motivated us to introduce in this paper an adaptive steerable total variation regularizer (ASTV) based on geometric moments. The proposed ASTV regularizer is capable of denoising the edges based on their geometrical orientation, thus boosting the denoising performance. Further, earlier works exploited the sparsity of the natural images in DCT and wavelet domains which help in improving the denoising performance. Based on this observation, we introduce the sparsity of an image in orthogonal moment domain, in particular, the Tchebichef moment. We propose a new sparse regularizer based on the Tchebichef moment. The overall denoising framework is optimized using split Bregman-based multivariable minimization technique. Experimental results demonstrate the competitiveness of the proposed method with the existing ones in terms of both the objective and subjective image qualities.

"Fusion of Deep-learning-based Descriptors and Human Memory Model for Weakly Supervised Content-Based Image Retrieval in Large-Scale Datasets"

Farzad Sabahi,

Ph.D. Candidate, Dept. of Electrical and Computer Engineering
Concordia University

Bio: Mr. Sabahi is a PhD Candidate in field of electrical and computer engineering at Concordia University, Montreal. His research has focused on pattern recognition and image retrieval. He has more than 20 refereed publications**.**

Abstract: The advent of low-cost digital recording and storage devices as well as the rapidly increasing popularity of online social networks and applications involved with large-scale image datasets make extended use of visual information. Therefore, interpreting images semantically and image retrieval have gained considerable attention among researchers. Therefore, not surprisingly, a lot of companies across various industries are looking to utilize advanced computational techniques to find useful data hidden across a set of huge visual data. One of the techniques commanding a lot of attention is deep learning. But, as a matter of fact, most of studies in the field of image retrieval have focused on sophisticated methods to employ expensive deep learning algorithms. Even though they have shown significant performance, most of current deep-learning-based methods in the field of image retrieval are impractical because they require a large amount of data for training without which the methods are highly unlikely to outperform other available approaches. The main objective of this research is to develop a multi-layered content-based image retrieval by incorporating deep learning and human memory models, to search among huge dataset effectively and efficiently to overcome the aforementioned disadvantages through hierarchical learning process. The other contribution of this research would be proposing a hierarchical neural network to take spatial features as well as semantic information into account to leverage huge amount of weakly indexed data. Since not known empirical research has focused on exploring relationships between deep learning and memory models, the proposed algorithm may be useful in various applications that contribute to the new era of multidisciplinary artificial intelligence and computer vision that is used in interpreting the images semantically such as medical image processing or internet of thing.

"Simultaneous Compression of ECG, PPG and Respiratory Signals and its Application in Web-based Remote Healthcare Systems"

Dr. Sourav Kumar Mukhopadhyay,

Postdoctoral Fellow, Dept. of Electrical and Computer Engineering
Concordia University

Bio: Dr. Sourav Kumar Mukhopadhyay received the B.Sc. (Hons.) degree in Electronics in 2005, M.Sc. degree in Electronic Science in 2007, and Ph.D. degree in Applied Physics (Instrumentation Engineering) in 2015 from the University of Calcutta, Kolkata, India. From 2008 to 2015, he was a part-time teacher in the Department of Electronics, Acharya Prafulla Chandra College, Kolkata, India. He is currently a Postdoctoral Fellow in the Department of Electrical and Computer Engineering at the Concordia University, Montreal, QC, Canada. He is a recipient of the Regroupement stratégique en microsystèmes du Québec (ReSMiQ) postdoctoral scholarship, February, 2017. His research interests include biomedical signal processing and analysis, bio-signal compression, feature extraction from bio-signals and disease classification. He has served as guest editor, editorial board member and reviewer of several international journals and major conferences. He has published 11 papers in various reputed high-impact International journals, and 11 papers in various conferences to date.

Abstract: Advancements in electronics and miniaturized device fabrication technologies have enabled simultaneous acquisition of multiple biosignals (MBioSigs), but the area of compression of MBioSigs remains unexplored to date. This research-work presents a robust singular value decomposition (SVD) and American standard code for information interchange (ASCII) character encoding-based algorithm for compression of MBioSigs for the first time to the best of our knowledge. At the preprocessing stage, MBioSigs are denoised, down sampled and then transformed to a 2D data array. SVD of the 2D array is carried out and the dimensionality of the singular values is reduced. The resulting matrix is then compressed by a lossless ASCII character encoding-based technique. The compressed file is then uploaded to a hypertext preprocessor (PHP)-based website for remote monitoring application. Evaluation results show that the proposed algorithm provides a good compression performance; in particular, the mean-opinion-score of the reconstructed signal falls under the category 'very good' as per the gold standard subjective measure. In 2015, as per the data presented at the American Epilepsy Society's annual meeting held in Philadelphia, the accuracy of seizure detection can be enhanced using MBioSigs. However, biosignals are subjective and hence, symptoms of physical and cardiac disorders may appear randomly. Therefore, the monitoring of biosignals has to be carried out for prolonged periods as done in the Holter monitoring systems (24 h). Consequently, the amount of data to be handled becomes enormous. An effective biosignal compression technique could be a solution to this highly growing rate of storage demand. Electrocardiogram (ECG) is considered as the prime and most important medical signal and Photoplethysmogram (PPG) is as the second most important one. Hence, these two signals along with the respiratory signal are chosen for the development of the compression algorithm for MBioSigs. Motivation behind the development of the proposed MBioSigs compression and tele-healthcare system is to produce a coherent snapshot of the patient to make it easier for the remote healthcare professionals to reach a decision and to spread the healthcare facilities beyond the hospital boundary. ECG signals often get heavily corrupted by various sources of high and low frequency noises. Hence, the task of noise elimination should be performed before any other intense processing. A discrete wavelet transform-based technique is used to denoise the ECG signal, and the PPG and respiratory signals are de-noised through a 4th order zero-phase bandpass filter. Denoised signals are down-sampled, and are rearranged to form a 2D array (denoted as Sigr x c). SVD of the Sigr x c is carried out, and the relatively small singular values are discarded. Finally the truncated matrix is compressed using the lossless ASCII character encoding-based technique. A website is developed based on PHP for secure storage and tele-healthcare application. At the patient's site, after compression, the administrator can upload the compressed file to the website, and assign doctor(s) for that particular file. Upon assignment, auto-generated emails are delivered immediately to the doctors requesting them to check and diagnose those signals. The proposed algorithm offers an attractive performance and its use could be particularly beneficial in hospitals or clinics, where different bio-signals are collected from thousands of patients every day. Another significant advantage of using this technique is that after compression time-domain biosignals are encoded into ASCII characters which inherently encrypts and increases the information security.

"Adaptive Real-Time Feature Extraction and Matching"

Muhammad Reza Pourshahabi,

Ph.D. Student, Dept. of Electrical and Computer Engineering
Concordia University

Bio: Muhammad Reza Pourshahabi has been a Ph.D. researcher student in Electrical and Computer Engineering at Concordia University under the supervision of Dr. M. N. S. Swamy and Dr. M. Omair Ahmad since September 2017. He was a full-time faculty member since 2009 till 2017 at the faculty of Computer Engineering, at Salman Institute of Higher Education, Mashhad, Iran. He received his M. Sc. in Computer Software Engineering in 2009 form Ferdowsi University of Mashhad, Iran. His thesis was on development of a Fully Automated PCO Quantification System based on Contourlet Transform in the field of Medical Image Processing. He earned his B. Sc. in Computer Hardware Engineering in 2006 form Ferdowsi University of Mashhad, Iran. His thesis was focused on Handwritten Signature Identification and Verification in the field of Digital Image Processing and Biometrics. His research interests are Feature Extraction and Matching, 3D Reconstruction, Multi Resolution Transforms and Biometrics.

Abstract: Feature matching is the primary and fundamental step in a vast area of vision-based applications. Some of these include image registration, object recognition, image retrieval, target tracking, stereo matching, surface matching, image stitching, medical imaging, visual SLAM and many others. It is obvious that accuracy and efficiency of the feature matching process is crucial for the whole proficiency of these applications. An efficient feature matching relies on salient feature detector and also robust feature descriptor. Salient features are those which are invariant to geometrical and photometrical transformations such as scale, translation, rotation, affine and illumination, across some range of noise, distortion, viewpoint changes and so on. These features should be repeatable, distinctive and local. The repeatability property of these features may be the most important ones. The optimal number of detected features is also a challenging problem. A Robust feature descriptor describes each of the detected features uniquely in a form of a vector. Matching Features across different images with overlap scenes are found by comparing these feature vectors. Feature extraction and matching is a popular topic in recent decades and many algorithms were proposed, each of which has some drawbacks and advantages. In this research a summary of some popular algorithms is presented. These algorithms are Moravec's detector, Harris detector, Harris-Laplace, Hessian, Hessian Laplace, Hessian affine, Laplacian of Gaussian (LoG), Scale Invariant Feature Transform (SIFT), PCA-SIFT, Speeded Up Robust Features (SURF), Feature from Accelerated Segment Test (FAST), Binary Robust Independent Elementary Features (BRIEF), Oriented FAST and rotated BRIEF (ORB), Binary Robust Invariant Scalable Keypoints (BRISK), Fast Retina keypoint (FREAK). There are some performance measures for the evaluation of different feature extraction and matching algorithms. The most popular ones such as 1-precision, recall, putative match ratio, matching score, and entropy are reviewed in this research. There are also different datasets for performance evaluation of different methods. The standard Oxford dataset and Stanford Mobile Visual Search (SMVS) are two well-known ones. The research have been carried out till now about performance comparison of different methods are also presented. Feature extraction and matching process is a computationally intensive task. Floating point and complex computations such as square root, division, multiplication, and exponential operations are not desirable in real-time applications, on the other hand, with the increasing number of mobile applications which are based on feature extraction and matching and also the limitation of low computational power property of hand-held devices, the need for real-time and robust feature detector and descriptor is mandatory. The goal of this research is to suggest an adaptive feature detector and binary feature descriptor for real-time applications. The proposed methodologies are as follows: 1. A two-step feature extraction and matching algorithm will be proposed in this research, for real time applications, such as visual SLAM, in order to increase the speed and accuracy. 2. We must avoid complex computations, for real time feature extraction. It is desirable to do feature extraction only by using parallel add and subtract operations. In this research the possibility of using integral image to approximate Difference of Gaussian (DoG) will be investigated. 3. It is mandatory to use binary descriptors for real time applications to benefit from Hamming distance resulted from simple and quick XOR operation. A new binary feature descriptor is proposed in this research.

"Introduction of Adaptive Directivity Controlling Parameter for First-Order Steerable Differential Microphone Arrays"

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Bio: Ali Sarafnia received his M.S. (2012) from Sharif University of Technology in electronics engineering. In 2012, he joined Huawei Corp as a core back office engineer in network optimization center and worked on MSC of GSM network. He is currently working on his Ph.D. in electrical engineering at Concordia University., Montreal, Quebec. His research interests include signal processing, speech processing, specifically speech enhancement techniques, design of microphone arrays for speech enhancement applications and Bayesian filtering for noise reduction purposes.

Abstract: In the real world, noise can impair both the quality and the intelligibility of speech signals. To reduce the effect of diffuse noise and reverberation interference, one can try to design a robust set in terms of digital signal processing algorithms and hardware parts of electroacoustic devices, such as microphone arrays. Microphone arrays are employed widely in the purpose of de-reverberation, sound localization, and noise reduction of the speech signal. This proposal tries to suggest an adaptive method for optimization of the directivity controlling parameter (α) of differential microphone arrays (DMAs) in the sense that this parameter changes adaptively to steer the beam pattern of the microphone array in the desired direction. In other words, an innovative version of the first-order steerable differential array (FOSDA), which is adaptive, will be introduced. Interestingly and simply, the adaptive FOSDA lets a speaker move freely in a room while it tracks a speaker's location and the beam direction sets electronically to its corresponding angle. Modeling time difference of arrival (TDOA) from speaker to channels of microphone array by Gaussian Mixture Model (GMM), estimating the azimuth angle of the sound source location by employing Kalman filter, its extensions, or other Bayesian estimators, and utilizing this estimated angle to find the value of the directivity controlling parameter lead to steering the beam response of the first-order azimuthal steerable microphone array adaptively. This adaptive algorithm will give rise to de-reverberation, noise/interfere point suppression, better SNR values and an optimized directivity factor.

"Continuous Software Integration for Robotic Swarms and IoT Devices"

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Polytechnique Montreal

Bio: Vivek Shankar Varadharajan is a junior robotics researcher delving into the robotics literature, to discover intriguing facts and expand the knowledge base in due course. In 2015, he received his M.Sc. degree in Automation and Robotics from Technical University of Dortmund (Dortmund, Germany). He received his bachelor's degree in Electronics and Instrumentation Engineering from Anna University (India) in 2012. At present, he is a Ph.D. student, working with robotic swarms, the focus of his research is to robustly deploy software on to a swarm of robots (like Unmanned Aerial Vehicle groups) during a mission, to allow hot swapping of the onboard software. He is currently pursuing his PhD. At MIST Labs, École Polytechnique de Montréal (Montréal, QC, Canada). His research interests include distributed robotics, multi-robot systems, machine learning, artificial intelligence and Cyber-Physical systems.

Abstract: The currently available methods to deploy software to a group of Internet-of-Things (IoT) devices/robotic swarms, are rather primitive, and require physical access to the hardware. With the growing numbers of devices introduced by automation and the Internet-of-Things, there is a novel interest in methods and tools to deploy new code to active IoT devices, sensor arrays and swarms of robots. In particular, methods that integrate code modifications to a group of IoT devices and swarm of robots during a mission, with minimal interruption time, can grant longer autonomy, add flexibility to the system, and streamline follow-up missions without the need for device recovery. In this work we designed an Over-The-Air update toolset for a groups of IoT devices/robotic swarms to propagate code updates while in operation, without interrupting the mission. New update releases are generated as patches to the deployed code, and a consensus mechanism borrowed from swarm intelligence ensures the execution of a unique code version in the whole swarm at all times. Simulations were conducted with thousands of units to study the scalability and bandwidth consumption during the update process. Real deployment experiments were then performed on a small swarm of commercial quadcopters that demonstrated the effectiveness of the tool.

"Measurement-Based Handover Method for Communication-Based Train Control Systems"

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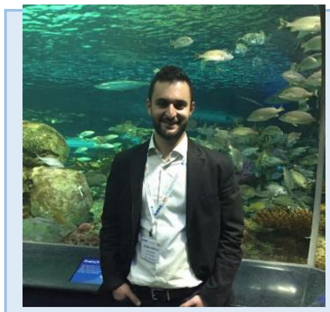
Postdoctoral Fellow,
University of Toronto

Bio: Sami Baroudi received, respectively in 2007, 2009, and 2016, a B.Eng. degree in Electrical Engineering from University of Balamand, Lebanon, an M.Eng. degree in Electrical Engineering, and a PhD degree in Electrical Engineering from Concordia University, Canada. He is currently a Postdoctoral Research Fellow in the department of Electrical and Computer Engineering at University of Toronto, in the research group of Prof. Jorg Liebeherr. His research interests include Communication-Based Train Control (CBTC) systems and radio propagation modeling specially for tunnels.

Abstract: In Communication-Based Train Control (CBTC) systems, trains communicate with wayside access points (APs) placed along a track using 802.11 or similar wireless technologies. A critical component of the communication system is the handover algorithm used by a train to select an AP as it moves along a track. As a safety-critical system, the number of handovers should be kept small and selected APs must satisfy given QoS requirements. We present a novel measurement-based handover algorithm for CBTC systems that exploits the predictable motion paths of trains. Using empirical measurements from a rail communication system, we show that our algorithm keeps the number of required handovers small, while avoiding ping-pong handovers, i.e., situations where a train bounces back and forth between the same access points.

Note that this is an abstract of an accepted paper in VTC Fall 2017.

"Modular Multilevel Converters for Flexible Integration of Renewable Energy Resources into Smart Grids"



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Bio: Mohammad Sleiman (S'12) received the B.Sc. degree in Electrical Engineering from the Lebanese University and the M.Sc. degree in Renewable Energy from Saint-Joseph University and Lebanese University, Beirut, Lebanon, in 2012. Currently, he is a PhD Candidate at the École de Technologie Supérieure, working on Control of Modular Multilevel Converters under the supervision of Prof. Kamal Al-Haddad. His research interests includes modeling and control of Multilevel and Modular Multilevel Converters for renewable energy applications and predictive control of power converters. He serves as a reviewer for several IEEE journals and major conferences.

Abstract: Modular Multilevel Converters (MMC) are known for their modularity, high efficiency, and reliability. Such features are highly demanded in today and future smart electric grids. As we move towards more and more industrialized societies, our dependence on electric power will continue to increase, with a great focus on stable, diverse and reliable type of power networks. Power downtime for today's societies is very costly and hence, is unacceptable. The MMC in this sense presents a multidisciplinary solution that can meet future electric power expectations, in terms of flexible integration of renewable energy resources with the smart grids. Due to the superior features of Modular Multilevel Converters, big market players in power generation have adopted the MMC as a reliable and effective high-power converter for smart grids. The increased popularity of Electric Vehicles (EVs) requires an efficient, reliable and a flexible power conversion scheme that allows a smart integration of EVs in smart grids. In addition, the MMC could be used to solve challenges related to the intermittent nature of renewable energy resources by including battery energy storage systems within the converter modules or (cells). The modular and flexible structure of the MMC allows EV batteries with different initial State-of-Charge to be smoothly used in charging and grid support (vehicle-to-grid) functionalities. Moreover, the need for energy storage systems is also increasing along with the increased penetration of distributed renewable energy resources. Therefore, the MMC as a versatile power converter structure with its battery storage integration capability meets the needs of today's and future smart grid requirements. Due to its attractive features, the MMC became a hot research topic worldwide. For this reason, authors decided to build two MMC down-scaled three-phase, 6kVA lab prototype for academic and industrial research and development purposes. The built MMC experimental setup allows authors to test and validate different control strategies that fits with the scope of smart converters, which interfaces distributed renewable energy resources with smart grids. Moreover, the converter topology used in this project uses 60 power cells in each MMC system and serves as an AC-DC bidirectional converter with superior features in terms of, reliability, modularity, efficiency, availability, faults handling capability, and multilevel voltage wave form (i.e. excellent power quality at the input and output ports). These superior features of the MMC, as compared to classical converters, permits ease of scalability in terms of voltage and power. The MMC system presented in this project targets renewable energy applications and its flexible connectivity to the AC or DC electric networks for more reliable and efficient smart grids. Such converter structure (MMC) fits a wide range of operations, including: - Power quality converters that can optimally fit in medium to high voltage smart grids, including active harmonic filtering (Active Filters) and reactive grid support capabilities (as in STATCOMS). - High power inverters that interfaces photovoltaic, onshore, and offshore wind farms to the smart grid. - Fast and Bidirectional battery charges/Inverters for EVs applications. - Efficient and reliable high power motor drives.