2nd DOCTORAL CONGRESS in ENGINEERING

8 – 9 JUNE 2017 • FEUP • PORTO • PORTUGAL

Book of Abstracts

2nd Symposium on Electrical and Computers Engineering

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Book of Abstracts

of the

2nd Symposium Electrical and Computers Engineering

Editors:
Aníbal Matos, Vítor A. Morais, Vítor H. Pinto

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Preface

The Doctoral Programme in Electrical and Computer Engineering (PDEEC) is offered at FEUP by the Department of Electrical and Computer Engineering (DEEC). The programme has close relations with INESC-Porto, ISR-Porto and INEB. Joint programmes also exist between DEEC and MIT, Carnegie-Mellow University and the University of Texas at Austin.

As it happened before, in this 2nd Edition of the Doctoral Congress on Engineering, which will take place at FEUP on 8th and 9th of June 2017, there will be a Symposium on Electrical and Computer Engineering. Since the Programme is multi-disciplinary and gathers several engineering areas, PDEEC invites all students and researchers to submit an abstract within the following areas:

- Power Electronics
- Electrical Power Systems
- Systems and Control
- Communications and Networks
- Signal Processing
- Image Recognition
- Machine Learning
- Microelectronics
- Robotics
- Operations Research
- Artificial Intelligence
- Bio-engineering
- Multimedia
- Real-Time Systems
- Other Topics in Electrical and Computer Engineering

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Programme

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Advanced functionalities for the future Smart Secondary Substation

Konstantinos Kotsalos[1;2], Nuno Silva[2], Helder Leite[1]

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Author Keywords. Low Voltage Network, Secondary Substation, Smart Grid Implementation.

Introduction

The continuous growth of multiple distributed generation (DG) and generally Distributed Energy Resources (DER) along the distribution networks induces multifaceted technical challenges on the distribution network operation. This work focuses on proposing novel multi-temporal operation scheme for LVDN based on advanced management functionalities, accommodated on a secondary substation housed controller entitled Distribution Transformer Controller (DTC). The main concern is to take advantage of responsive DERs such as controllable loads of residential users under demand side management (DSM) actions, battery storage systems and microgeneration units in order to coordinate the operation of LVDN. The goal of this control scheme is to ensure bilateral benefits for end-consumers' and Distribution System Operator (DSO) as well as to guarantee that the overall technical constraints are being respected.

Conceptual architecture and Control-scheme

The LV grid generally is comprised of a large scale dimension with a significant number of buses, i.e. hundreds to thousands of buses where the observability of its state or even the topology is often unavailable to the DSO. Nevertheless, the latest years the breakthroughs occurred merely on the frame of Information and Communication Technologies (ICT) and the automation of electrical grids, contributed significantly to gradually change the land-scape on LV grid's observability, control and management functionalities. The extensive installation of Advance Metering Infrastructures, that most of DSOs are investing on; fact which enables the capability to acquire analytical measurements from end-users in a real-time basis (Olival et al., 2017).

Concurrently, the enduring integration of renewable sources and generally micro-generation sources at the LV level possibly poses some technical challenges related to the operation of the network itself.

More than a significant technical challenge, it is an evolutionary opportunity to progress towards the Smart grid concept integrating advanced ICT, in addition to distribution and substation automation to ensure a proper interface between DGs and the electric utilities.

The major goal of the research project is to propose control schemes providing advanced monitoring and control functionalities considering coordinated operation of storage systems, loads under DSM and dispatchable microgeneration units. The approach comprises

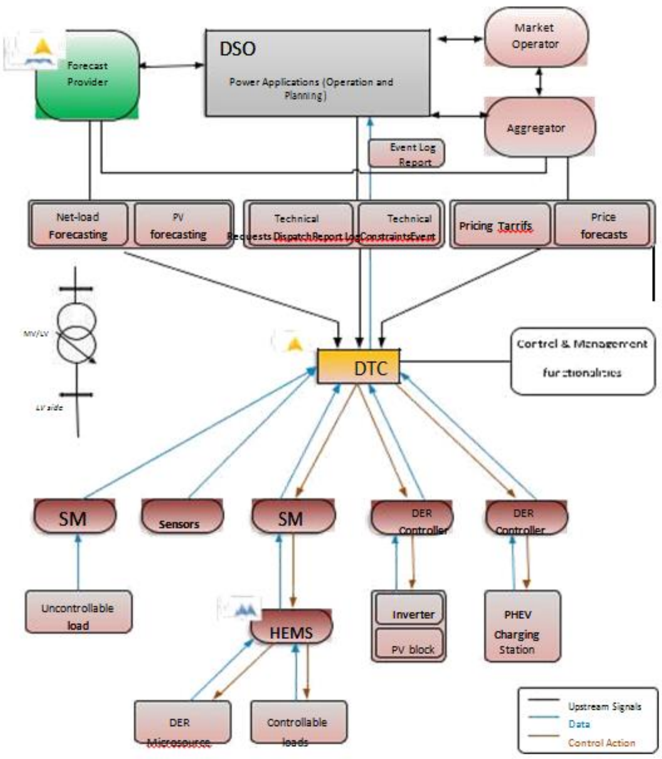

Figure 1: Conceptual architecture for the LVDN operation.

of a day-ahead operational planning that will be cyclically adjusted every single hour. The control scheme will be accommodated on the "substation-centered" controller, the DTC. The overall control scheme will intend to make use of the flexibility services of the several DER connected devices. Hence, the DTC acts as an aggregator of these flexibilities through the Smart Meters (SM) and Home Energy Management System (HEMS) (Fig. 1). Therefore, a bottom-up flexibility aggregation will take place catering for the participation of the residential consumers and generally the DERs. Then, the flexibility data can be exploited to resolve the planning of LVDN day-ahead operation, while the flexibilities could either be transmitted to the wholesale market through the delegate entity of aggregator. The existence of the aggregator intends to procure the right to control the demand pattern of the flexible loads in exchange of compensation fees.

For the overall scheme, there is particular need to rely on short-term demand and renewable generation forecasting, so as to define the control actions for the following day and hours, concerning the day ahead and hourly based scheme respectively.

An important monitoring issue within LV grid in real field is the observability, despite the availability of a set of measures with sufficient redundancy possibly allow to obtain the current state of the grid, by determining unobservable state variable in real-time (Miranda et al., 2012). The availability of such sets of measures might be acquired by SM (Chen-Ching Liu, 2016).

Concerning proposed scheme, the control actions are based on control set-points that are sent by the DTC to the smart meters and smart inverters, to implement the decided tuning. Inputs in the controller through the control center are day-ahead forecasts for the microgeneration and the daily load profile as well as technical set-points to control the power flow downstream to the substation. State variable in real-time.

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Integration of Variable Speed Pumped Hydro Storage in Automatic Generation Control Systems

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Author Keywords. Variable-Speed Hydro Pumped Storage Power, Automatic Generation Control, Frequency Restoration Reserve.

Extended Abstract

Although their acknowledged advantages, generation from new renewable energy (NRE) induce high levels of uncertainty and variability in the system when compared to the conventional technologies. To overcome these major drawbacks, energy storage devices have been referred as a possible solution that is able to leverage high levels of NRE deployment (Cheng, Tabrizi, Sahni, Povedano, & Nichols, 2014). Particularly, along with these are the pumped storage power (PSP) plants, which explore the possibility of using reversible units, able to either generate power or to pump water to an upstream reservoir. PSP plants have proved to be the most utilized, mature, efficient and cost-effective technology for large-scale energy storage, and may be playing and important role in handling the NRE integration issues in the future by providing storage capacity and advanced system support function in the form of fast ramps for power balancing purposes. The need for generation units' increased operational flexibility has driven the exploitation of new power generation solutions within the hydropower context by exploring variable-speed machines. The controllability provided by converter-connected units enables faster power-to-grid responses through the exploitation of higher ranges of turbines' speed and respective kinetic energy stored in its rotating masses, since the generator rotational speed is decoupled from the grid frequency. However, the massive integration of NRE and subsequent replacement of conventional machines have significant impacts on the frequency control robustness either due to the enormous reduction of the system inertia as well as because of the reduction of the number of fully controllable units connected to the grid.

In order to provide grid frequency control services, generation units are required to provide enough power reserves. Besides the automatic and local frequency containment reserve (FCR), generating units must also provide frequency restoration reserve (FRR) which is activated centrally by the system operators through automatic generation control (AGC)

systems, responsible for the re-establishment of the nominal system frequency and the power interchanges between the so-called control areas. In line with this control

organization in electric power systems, this paper presents and discuss the role of a variable speed hydro PSP plant in the regulation mechanism supported by the AGC system, by comparing the performance of a variable speed unit with a conventional synchronous-based machine. In order to deliver these studies, a reduced-order model of a hydro PSP plant was previously developed (Moreira, Fulgêncio, Silva, Nicolet, & Béguin, 2017) where the hydraulic part of the unit was derived and validated by comparison with a full-detail hydraulic setup.

Test case and results

A three-area network test system, developed in MATLAB/Simulink®, was used in the performed simulation studies, comprising three main voltage levels: 21,6kV for the generation systems, being the transmission grid established at high and extra-high voltage levels of 230kV and 500kV respectively. It is assumed that one of the generators consists of a hydro PSP while all the other generators are referred to thermal units using synchronous machines. To study the response of the hydro PSP plant within the AGC scheme, a 50 MW load increase in Area 3 (where the hydro PSP plant is located) was simulated. A conventional synchronous-based hydro unit performance was compared to a variable-speed unit using a doubly-fed induction machine (DFIM). In figure 1, it is possible to observe the tie-lines power flow (Ptie12 and Ptie23) re-establishment, in the moments subsequent to the disturbance occurred in Area 3, is significantly faster when using the variable-speed technology. Its controllability and improved power-frequency response characteristics leads to a faster tracking capability of the AGC control signal and consequently faster response when re-establishing of tie-line power flows. To support this power-to-grid response, as it can be observed in figure 2, the unit takes advantage of the wider range of available turbine speed. When the load increase occurs the power converters respond almost immediately and the speed deviation around the nominal value tends to be much larger. Additionally, the frequency recovery occurs also faster for the variable-speed unit due to the same reasons.

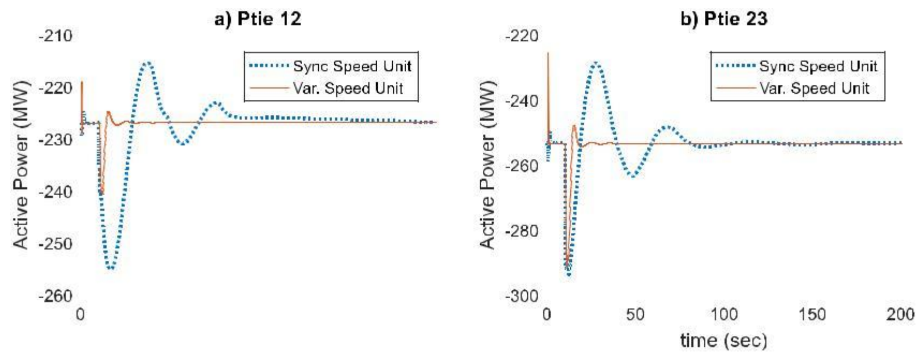

Figure 1: Tie-lines power flow compared responses to 50MW load increase in Area 3, for synchronous and variable-speed hydro PSP plant

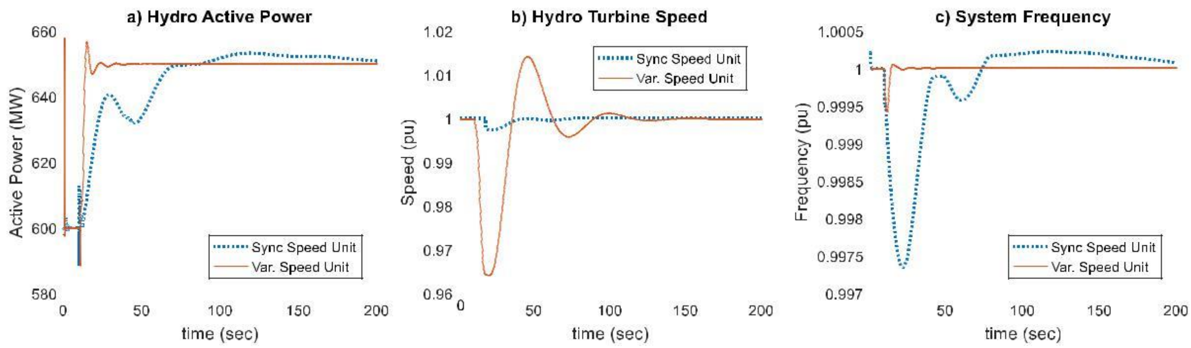

Figure 2: Synchronous and variable speed hydro PSP plant response to 50MW load increase in Area 3

Conclusions

Pumped storage power (PSP) plants are expected to play an important role when dealing with NRE integration by accommodating some of its uncertainty and variability. However, the replacement of conventional synchronous units for NRE – which normally rely on power-electronics connections to the grid – significantly decreases the system inertia and, hence, the robustness of frequency control. In order to understand how can PSP plants participate in the frequency restoration reserve (FRR) provision, a three-areas network controlled by an automatic generation control (AGC) system, was implemented and tested in this paper. These units have proven to be much faster when compared to conventional units, responding positively to disturbances and in the provision of FRR. Such conclusions demonstrated variable speed PSP ability to provide key system balancing functionalities in scenarios with high shares of NRE.

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Comparative analysis of visual odometry implementations

Alexandra Nunes[1], Ana Rita Gaspar[2], Andry Pinto[3], Aníbal Matos[4]

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Author Keywords. Evaluation, monocular, odometry, vision

The use of visual odometry methods in autonomous vehicles has been growing, allowing to estimate the egomotion while receiving new visual data. With the use of optical sensors, the vehicle uses the features of the environment to accomplish this task. These acquisition systems have an appellative cost and supply a larger quantity of data. In this context, the paper presents a comparative analysis of three visual monocular odometry implementations, since it's important to select the most suitable considering the wide variety of robotic applications, facilitating future choices for different tasks that require robot's motion.

Introduction

Considering the high number of monocular visual odometry methods available in the literature, it's necessary to analyze the performance and the advantages and disadvantages of each implementation, in similar scenarios. Thus, the main properties of each implementation were identified to choose the most appropriate for the task to be accomplished. Therefore, was analyzed the performance of three implementations, using datasets avail-able online, to give the most appropriate without forgetting the applicability in a real robotic system. After the literature review, the mono-vo, viso2 and ORB-SLAM were studied.

Materials and Methods

According to the literature review, each implementation presents important particularities. The mono-vo is simple, based on OpenCV and presents a method of removing outliers. The viso2 is very configurable, uses bucketing and requires the camera to be at a fixed and known height from the ground, causing errors with pure rotations.

Finally, ORB-SLAM presents a keyframe-based approach for motion estimation, uses the same features for all steps (most efficient and simple system) and offline vocabulary for motion estimation and relocalization. In order to analyze the performance of the implementations was used, for example, the KITTI dataset (urban environment), as shown in Figure 1, considering the normalized Euclidean error in motion estimation, CPU utilization and processing time.

Figure 1: Illustrative example of KITTI dataset

Discussion

According to the comparative analysis, it was verified that the ORB-SLAM implementation was the best in the motion estimation. However, this implementation uses an offline vocabulary provided by the authors, who makes it difficult to adapt to different scenarios. Although it is a SLAM implementation, only the motion estimation was verified without areas revisited. In terms of processing time and computational requirements, the viso2 implementation was the best. This presents good estimation results, excluding situations where the height of the camera changes in relation to the ground or in the presence of abrupt and pure rotations. Although the mono-vo tries to replicate the motion of the camera, its heuristic becomes a limitation when there are more complex trajectories, since it allows only forward motion.

In Table 1 it’s possible to summarize the characteristics evaluated.

	Mono-vo	**Viso2**	**mORB-SLAM**
Mean error	-	+	++
Processing time	+	++	-
CPU	-	++	+

Table 1: Performance of the implementations with the evaluated parameters

Conclusions

From the analysis, it was clear that the ORB-SLAM (Mur-Artal Raul, J. M. M. Montiel and Juan D. Tardós 2015) and viso2 (Geiger, Ziegler and Stiller 2011) implementations are, in general, the most appropriate. On the other hand, the demonstrated efficiency on the ORB-SLAM and the fact that the implementation viso2 is quite configurable are important aspects. In case there is no suitable vocabulary for the desired scenario, the implementation viso2 stands out. In this way, the study showed importance because it allows to obtain assertive conclusions of each implementation, when submitted to the same application context.

Acknowledgements

This study was one of the topics analyzed in the development of the MSc dissertation. The authors would like to thank FEUP and INESC TEC for the conditions created for its development.

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Comparative analysis of visual SLAM implementations

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Author Keywords. Evaluation, SLAM, stereo, vision.

The use of SLAM (Simultaneous Localization and Mapping) methods in autonomous vehicles has been growing for their navigation, allowing them to construct a map of the environment while they are located on that same map. With the use of optical sensors, the robot uses the features of the environment to accomplish this task. These acquisition systems have an appealing cost and supply a larger quantity of data. In this context, this paper presents a comparative analysis of three visual stereo SLAM implementations, as it is important to select the most complete ones taking into account the wide variety of robotic applications, which will help in future choices for different navigation tasks.

Introduction

Considering the high number of stereo visual SLAM methods available in the literature, it's important to analyze the performance as well the advantages and disadvantages of each implementation in similar scenarios. Thus, the main properties of each implementation were identified in order to choose the most appropriate for the task at hand. Therefore, the performance of three implementations was analyzed, using datasets available online, to estimate correct egomotion and check its application in a real robotic system. After the literature review, the ORB-SLAM2, RTAB-Map and S-PTAM were studied.

Materials and Methods

From the literature review, each implementation presents significant specificities. The RTAB-Map implementation highlights incremental loop detection, map optimization and the incorporation of libraries for motion estimation. S-PTAM uses binary features (faster extraction) and a keyframe-based approach for motion estimation. Finally, ORB-SLAM2 detects a loop from an offline vocabulary (available online) approach that is also used for motion estimation and relocalization. To analyze the performance of the implementations the KITTI dataset (urban environment) was used, as shown in Figure 1, considering the Euclidean error in motion estimation, CPU and RAM utilization and processing time.

Figure 1: Illustrative example of KITTI dataset

Discussion

According to the comparative analysis, it was verified that the ORB-SLAM2 (Mur-Artal, Montiel and Tardós 2005) implementation was the best in the motion estimation and computational requirements. In terms of processing time, the remaining implementations (RTAB-Map and S-PTAM) give better results, although they do not use all the frames for the motion estimation. This component becomes crucial due to the future applicability in the context of the real-time operation. However, S-PTAM (Pire, Fischer, Civera, Cristoforis and Berlles 2015) is the one with the highest computational requirements. It was possible to verify that only the ORB-SLAM2 is suitable for the recognition of revisited areas (closure loop detection), by discarding frames in the remaining algorithms. Table 1 shows a summary of the performance obtained by the different methods.

	RTAB-Map	**S-PTAM**	**ORB-SLAM2**
Median error	-	+	++
Processing time	+	++	-
CPU	+	-	++
RAM	+	++	-

Table 1: Performance of the implementations with the evaluated parameters

On the other hand, the behavior in ideal conditions (offline processing) was analyzed and, as expected, when the ratio between the processed frames and the acquired frames is higher, the performance is better. Over these conditions, RTAB-Map and S-PTAM significantly improved their behavior by detecting existing loops.

Conclusions

From the analysis, it becomes clear that the ORB-SLAM2 and S-PTAM implementations are, in general, the most complete. On the other hand, the S-PTAM emphasizes the minimization of the dependency between two threads and of the ORB-SLAM2 the efficiency and the effective proof of the quality of the loop detection.

The necessity to operate in real-time highlights the ORB-SLAM2 implementation, with the possibility of adapting the vocabulary to the context in question. This study is relevant because it obtains assertive conclusions about each implementation, when submitted to the same application context.

Acknowledgements

This study was one of the topics analyzed in the development of the MSc dissertation. The authors would like to thank FEUP and INESC TEC for the conditions created for its development.

References

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Towards Offline Real-World Network Experimentation using ns-3

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Author Keywords. ns-3, Mobile Network Simulation, Trace Based Simulations, Reproducibility of Experimental Conditions, Perpetuation of Real-World Mobile Testbeds, Offline Experimentation.

Extended Abstract

In the past years, we have been optimizing the development process of protocols for communications systems, mostly relying on Network Simulator 3 (ns-3). In the beginning, we only used ns-3 to create a simulator implementation of the solution, which was tested in multiple simulation scenarios. With the introduction of the ns-3 Emulation, we started using the Fast Prototyping of Network Protocols (Carneiro, Fontes and Ricardo 2011) methodology, which consists in reusing the same ns-3 code as a shared protocol model implementation between the simulation and the testbed (prototyping) environments, as illustrated in Figure 1. Later, we extended the compatibility of the ns-3 Emulation to support broader types of real-world networking scenarios (Fontes, Campos and Ricardo 2015), and improved its performance (Fontes, Cardoso and Ricardo 2016) with the goal of optimizing the fast prototyping of network protocols, represented by the left arrow in Figure 1. In the work presented herein, we explore how the simulation can benefit from the data gathered from the prototype (right arrow in Figure 1).

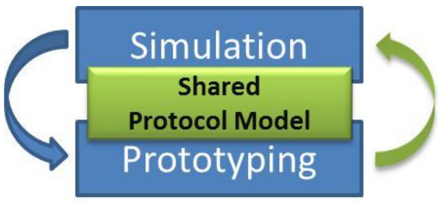

Figure 1: Simulation and Prototyping mutually beneficial relation via ns-3 Shared Protocol Model.

A common problem in mobile networking research and development is the cost related to deploying and running real-world mobile testbeds. Due to cost and operational constraints, these testbeds usually run for brief time periods but generate relevant results that are hard to reproduce.

Recording and Reproducing Real-World Mobile Network Testbeds

We propose the use of ns-3 as a solution to successfully reproduce real-world mobile testbed experiments. This is accomplished by feeding ns-3 with real testbed traces containing: 1) node positions, based on GPS coordinates; 2) radio link quality, based on the Signal-to-Noise Ratio (SNR), which is derived by the noise floor and the Received Signal Strength Indicator (RSSI) reported by the real interface. To reproduce such conditions in ns-3, a new ns-3 *PropagationLossModel* was developed, called *TraceBasedPropagationLossModel*, which can reproduce asymmetric radio link quality (e.g., different exposure to noise).

Preliminary Solution Validation and Conclusions

To validate our approach, the network throughput (TCP+UDP traffic) between a Fixed Base Station and an Unmanned Aerial Vehicle (UAV) was measured in a real-world testbed and recorded alongside with the real traces regarding node position and radio link quality (very asymmetric as UAV is more exposed to noise). The experimental results were compared to the network throughput (UDP only) achieved using the ns-3 trace-based simulation and a plain ns-3 simulation. The obtained results, presented in Figure 2, show that our trace-based simulation approach produced results much closer to the real scenario than using plain ns-3 simulation. Differences in maximum throughput are mainly due to the transport protocol used being different (TCP has higher overhead). The trace-based ns-3 simulation approach enables: 1) concurrent user access to the real testbed conditions based on past traces; 2) running simulations in faster than real time; 3) running multiple simulation instances at the same time, exploring different variants of the solution under evaluation.

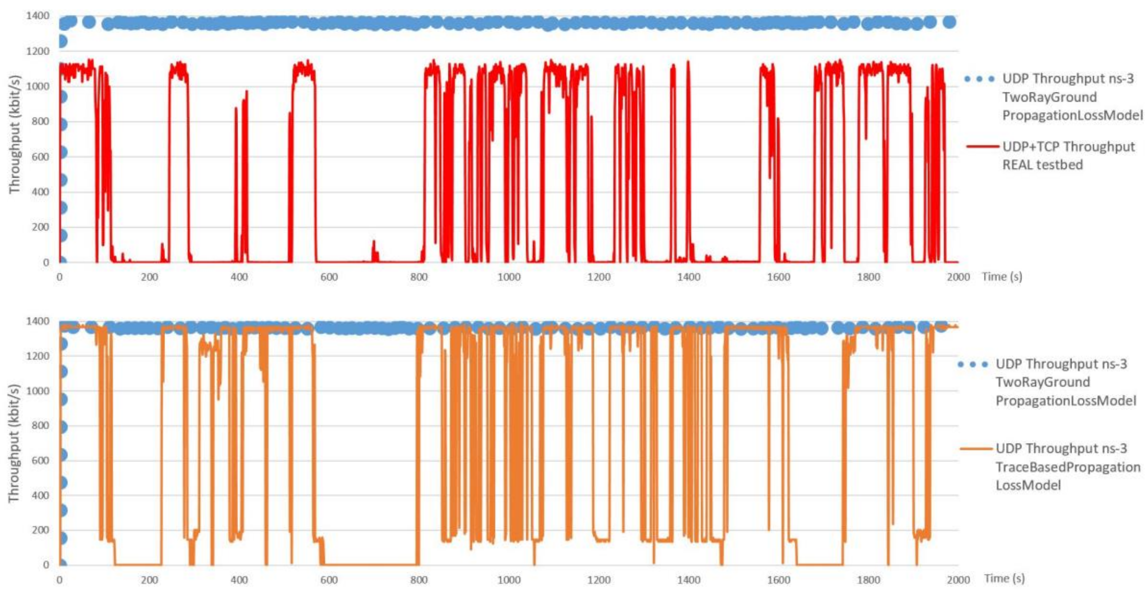

Figure 2: Real Throughput vs. Plain ns-3 Simulation vs. Trace-Based ns-3 Simulation.

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Power Amplifier Modeling using Neural Networks for Predistortion

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Author Keywords. Artificial Neural Networks, Predistortion, 5th Generation Networks

Extended Abstract

Power amplifiers are a part of wireless transmitters and their performance is critical in any transmitting chain. However, efficient power amplifiers are mostly nonlinear Teikari 2008. This is inconvenient because it generates two problems. The first one is that the nonlinear behavior affects both the amplitude and phase of the output signal. Since data modulation schemes used nowadays involve both amplitude (AM) and phase modulation (PM), the AM/AM and AM/PM distortion can cause errors to occur in the demodulation, and therefore, incorrect symbol detections. Added to these distortions, we must also take into account PM/PM and PM/AM distortions as shown in E. G. Lima2011. The second problem is related to bandwidth usage. Nonlinearities of the power amplifier spread the bandwidth of the output signal, which can cause interference in adjacent channels. This is even more important in emerging standards such as the 5th generation mobile communication system. This system will need to support larger bandwidths, in the order of 200 to 400 MHz. In addition, as the number of users is increasing, efficient use of the shared spectrum is necessary, which poses stringent demands on the linearity performance of the wireless transmitters.

To correct the nonlinearities of the power amplifier the most common way is to include a digital predistortion block in the system, before the power amplifier. This block produces the inverse of the transfer function (whichever it may be) of the power amplifier. It is important to note that the nonlinearity of the power amplifier is dynamic, which means that the predistortion method used needs to take memory into account.

The most common approaches on predistortion are based on look up tables, complex polynomials or, more recently, through artificial neural networks (ANN), and fuzzy systems Rexaei2013. The objective of this work is to model a power amplifier using an ANN with the purpose of later implementing an analog, low power predistortion block for 5th generation networks.

Examples of digital implementations of ANNs with predistortion purposes can be seen in N. Naskas2002 and Y. Qian2006. The need for using an analog predistortion block was imposed by the low power demands of 5th generation systems.

In our approach we use an activation function that can have an easy analog implementation.

The neural network created consists of four layers: the input layer, two hidden layers and the output layer. The inputs of the ANN are obtained through a tap delay line.

One of the optimization parameters of this ANN is the number of previous samples to be used, i.e. the number of inputs of the ANN. For instance if we create an ANN with 3 inputs, they will be the current input sample, and the 2 previous input samples. The other main optimization parameters explored are the learning rate, the number of neurons in each layer, as well as the activation function itself. The success of the neural network is evaluated based on the normalized mean square error (NMSE) of the output of the ANN on the test phase of the neural network. The samples used for the training of the network are from a gallium nitride power amplifier with data taken from the downlink from a mobile communication base station. About 40000 samples are used in the training of the network and 10000 samples are used for the testing. All tests were done numerically in a MatLab environment.

In Fig. 1 we can see spectral density of the results of an ANN used to model the inverse of a PA. In blue we have the output of the PA (which we use as input of the ANN), in yellow we can see the input of the PA (which we use as a target for the ANN) and in orange we have the output of the ANN. It is clear that the ANN corrected some of the distortion imposed by the PA.

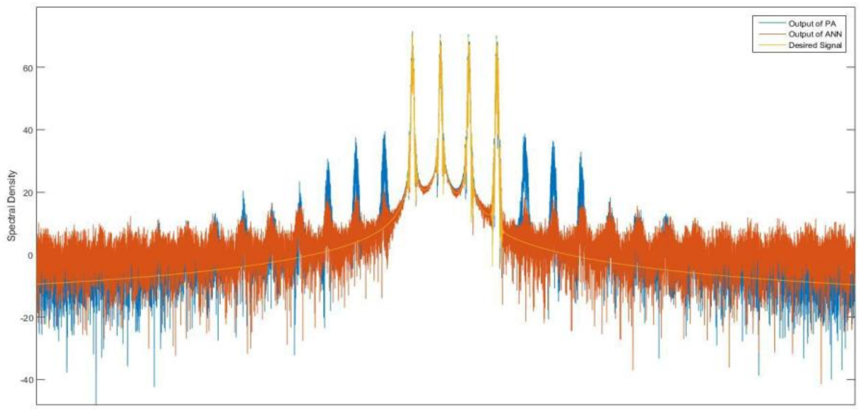

Figure 1: Results of an ANN used to model the inverse of a PA

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Development of Baseband Processor for Enhanced Data Rate Communication in TV White Spaces

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Author Keywords. Orthogonal Frequency Division Multiplexing, TV White Spaces, Wireless Personal Area Networks, Machine to Machine Communication, Field-Programmable Gate Array

Extended Abstract

In this work, a full custom baseband processor is demonstrated for high data rate communication. The deployment of digital video broadcasting (DVB) around the world led to unused sub-GHz frequencies (commonly known as TV white spaces – TVWS), which has been attracting several researchers to develop new radio frequency (RF) systems.

The authors propose a full digital processing system, intended for machine to machine (M2M) communication, which will exploit these selected channels, to attain better material penetration characteristics. Additionally, orthogonal frequency division multiplexing (OFDM) is employed to handle the multipath propagation, inherent to certain environments, like hilly terrains and indoor scenarios. Combined with the high spectral efficiency of OFDM, data rates in excess of 10Mbps can be achieved, even under heavy code rates, in a single 6MHz channel. Furthermore, it will be demonstrated that the bit rate can be further increased if non-contiguous OFDM (NC-OFDM) is used, when more than one channel is available. The complete system will be developed in hardware description language (HDL) and tested in a field-programmable gate array (FPGA).

Introduction

With the growth of the internet of things (IoT), new communication impairments emerged, leading to the development of new wireless personal area networks (WPAN) standards, such as IEEE 802.15.4. Despite fulfilling its role rather competently, some devices require higher data rates. Such is the case of real time image and video processing or dynamic environments, where a sufficiently high channel capacity is of the essence, mainly for M2M communication.

Many researchers managed to surpass this inconvenience either by exploiting 802.11 and mobile network infrastructures (specially due to its availability) or by relying in ultra-wideband (UWB) transceivers. However, the former offers an undesirable processing overhead, which translates in higher power consumption. On the other hand, UWB is incapable of offering a useful coverage area, when operating above 5GHz, and its current European specifications, at sub-GHz operation, has a very low power spectral density requisite.

In this work, an alternative solution is presented by reusing the, now unoccupied, old analog TV signal spectrum (distributed in selected 8 MHz channels from 54MHz to 790MHz).

By operating at sub-GHz frequencies, RF signals are offered better propagation characteristics, due to its inherent higher signal wavelength (even if line of sight is unattainable).

Work Developed

As a start point for this work, an analysis was made of typical channel models, in order to find the possible root mean squared (RMS) delay spread to which the RF signal will be subject. With this value in mind, the main system requirements for employing OFDM were calculated, which can be found in table 1. With all parameters selected, it can easily be proven that by applying high order amplitude and phase modulations, the 10Mbps boundary is achieved.

Expected RMS Delay Spread	Cyclic Prefix	OFDM Symbol Duration	Data Subcarriers	Required FFT Size
ϕў϶ϴ[illegible]	[illegible]	ĸJ[illegible]	ϕνϵ	ϵϴɛ

Table 1: Expected parameters of interest for OFDM Baseband Processor.

Since the system will be developed in HDL, fixed point calculations are required, as they offer the best compromise of occupied area and system complexity. A full qualitative and statistical analysis will be provided for all designed blocks, assuring a low output signal degradation. An example variation of the output error, for different word lengths, can be observed in Fig. 1, for a core block.

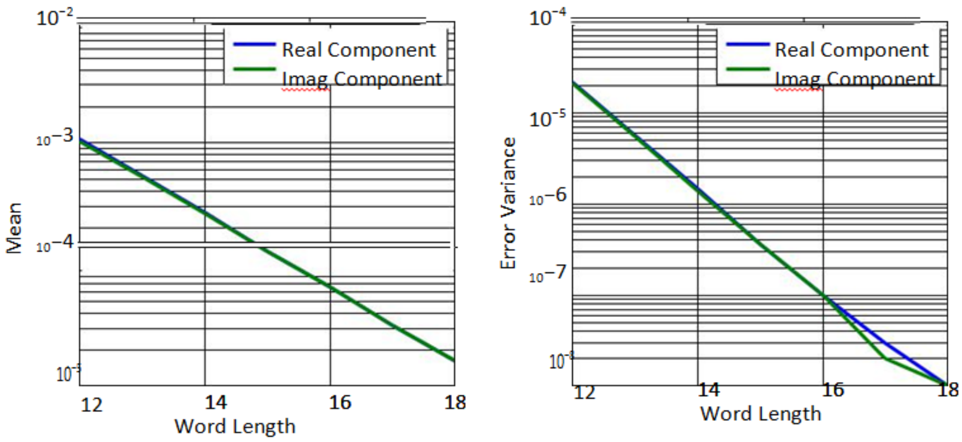

Figure 1: Coordinate Rotation Digital Computer (CORDIC) error variation with the increase of fixed point variables word length.

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Musically-Informed Adaptive Audio Reverberation

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Author Keywords. Audio Effects, Music Information Retrieval, Digital Signal Processing, Digital Reverberation, Musical Performance, Computational Creativity

We present a novel digital reverberation effect capable of adapting its output to the harmonic context of a live (i.e., real-time) music performance. The proposed reverberation effect is aware of the harmonic content of an audio input signal, and filters the resulting output accordingly. In other words, the signal outputted by the reverberation is "tuned" to the harmonic content of the performance. Conversely to the traditional (non-adaptive) reverberation effects, which emulate the physical phenomena of sound waves reflecting on enclosed space surfaces (Zölzer, 2011) — treating all input harmonic content equally — our implementation encompasses a dynamic behaviour, in the context of the recent research line of Adaptive Digital Audio Effect (A-DAFx) (Verfaille and Arfib, 2001), and thus avoids the sonic clutter typical of a traditional reverberation effect (Sterne, 2015).

The proposed adaptive reverberation is implemented as a live guitar pedal effect, which acts as control of the harmonicity level of the resulting output. When fully released only a few harmonic partials related to the input signal are allowed through the reverberation unit, and when is fully depressed the pedal acts as a traditional reverberation. Within these two extremes, a wide spectrum of creative possibilities is offered to the user. In providing musicians and composers with a content-aware audio reverberation effect whose controlled output blends with the harmonicity of an ongoing performance, we aim to foster new creative experiences.

We adopt the perceptually-inspired Tonal Interval Space as an audio-based representation of the harmonic content of an ongoing performance. From an input audio signal, we compute Harmonic Pitch Class Profile (HPCP) vectors (Gómez, 2006), in which the energy of the signal harmonic content is collapsed into a single octave. Afterwards, we compute Tonal Interval Vectors (TIVs), as the discrete Fourier transform (DFT) of HPCP vectors, whose components coefficients are further weighted by empirically consonance ratings (Bernardes et al., 2016). By distorting a the HPCP space using a weighted DFT, we make the Tonal Interval Space a perceptual relevant representation. In the resulting 12-dimensional space, Euclidean and angular distances between audio input TIVs indicate levels of perceptual proximity.

In other words, small distances between two sonorities express their levels of perceptual relatedness (e.g., the C note is closer to G then to B, even though the first interval is separated by 7 semitones and the second by 1 semitone).

We use the Tonal Interval Space as a framework to develop an algorithm that can efficiently identify, at given intervals of time, the harmonic "context" of an ongoing performance. After representing the audio input TIVs, we smooth their trajectory as the mean of the last 7 vectors to avoid sudden changes in the filter's behaviour. We then compute the distance of the 12 notes of the chromatic scale from the moving averaged TIV. The resulting distances are finally ranked in an increasing order of the perceptual proximity of the 12 chromatic notes from the input signal. The level of pedal depression is then used as a threshold to defines which pitches are allowed in the output reverberation.

The algorithms for audio signal analysis and resynthesis are implemented in Pure Data (Puckette, 1996). The choice of such programming environment was the possibility to easily port the code to the libPD library (Brinkmann et al., 2011), which we use within the BELA,a system dedicated to DSP processing with notable low audio processing latency. The BELA implementation is then used as an embedded processor for the guitar pedal. The use of the PD programming environment, and most notable the libPD library, also expands the application domains of the adaptive audio reverberation beyond the guitar pedal, as the library has been ported and widely adopted as a VST plug-in host in many digital music software for performance and audio production, such as Digital Audio Workstations and game engines (Enzien Audio Ltd., 2016).

In https://http://paginas.fe.up.pt/~ee12035/ we provide some initial results of the system for different performances and a controlled range of parameter settings.

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Interactive Manipulation of Musical Melody in Audio Recordings

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Author Keywords. Melody manipulation, melody detection, QT

Introduction

Melody is one the most defining characteristics of a piece of music, particularly those in the popular music genre. In this paper, we propose an algorithm to take the melody of a musical recording and change its components, modifying their pitch or deleting them altogether and then resynthesizing and playing back the results. A novel functionality of our system is to shift all the instances of a given note at the same time as well suggesting a set of possible pitch transformations for a given note. We seek accomplish not only the manipulation and transformation of melody, but also the development of an application that allows users to experiment with existing musical content regardless whether they have the technical background to understand the inner workings of the system. In other words, our work aims to provide an uncomplicated manner for music enthusiasts to explore a more active and creative listening experience.

Background

One of the critical aspects of this work is the extraction of melody from a musical recording to, then, manipulate it. To achieve this, several methods that extract pitch, and subsequently melody, have been proposed. The precursor to melody extraction is monophonic pitch estimation (also referred to as pitch tracking), which is strongly based on f0 estimation. Pitch shifting is commonly used for pitch correction of musical performances or for transposing a piece of music to a different key (Schörkhuber et al., 2013). A prime example of pitch shifting is Autotune, where recorded vocals are altered so that they become perfectly in tune with the rest of the music. Pitch shifting is highly relevant to this work and it has been shown recently that it can performed using the well-known constant Q transform (Brown, 1991).

The most widely used method for melody detection is the salience based method. This approach relies on the creation of a function which can identify pitch candidates in polyphonic signals. Although there are several different published variations on this approach, they consist of the same basic stages (Cancela, 2008; Salamon & Gómez, 2012). Another strategy is to separate the melody from the remainder of the mix, for instance the method proposed by Durrieu (2010) models the power spectrogram of the signal as a sum of the lead voice and accompaniment contributions.

Approach

In our work, we use the invertible constant Q transform (Schörkhuber & Klapuri, 2010) for experimenting with audio input signals. To this end, we compiled a dataset which comprises melodic excerpts from various musical tracks of several different music genres. Melody estimation is performed using the state of the art MELODIA Vamp plugin (Salamon & Gómez, 2012) for Sonic Visualizer. Here the input's main melody is automatically annotated through the identification of its fundamental frequency, and a melody activation function calculated using the salience based approach. Next, we use the MATLAB toolbox developed by Schörkhuber & Klapuri (2013) which includes an implementation of the invertible constant Q transform and contains a pitch shifting tool – normally used for global pitch shifts (i.e. to all notes at once). In our work, we integrate the extracted melody contour into the CQT representation and then pursue its manipulation via pitch shifting of individual notes in the CQT representation. This is achieved by quantizing the melody contour into the CQT, with a chosen number of bins. Next, the bins that contain melody and its harmonics that are automatically identified, to more accurately preserve its original characteristics when transform. Prior to pitch shifting, a mask is generated which isolates the main melody and its harmonics. The transformations to the melody are then done by selecting notes from the melody (along with their harmonics) and altering their pitch via the pitch shifting functionality in the CQT toolbox. Via an interactive interface, we allow users to specify which notes they wish to modify, as well as giving them the option to capture all instances of the same note, and providing suggestions as to which pitch shifts to apply to any user selected note.

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Interactive Content Selection Applied to Multimedia Applications

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Author Keywords. Human Computer Interaction, Computer Graphics, Content Selection, Multimedia Content, Intel RealSense

Extended Abstract

In today´s content-centric world, the aspects of usability and easiness in selecting desired content are essential to deliver successful interactive multimedia solutions. Set-top boxes, for instance, apart from their core functions such as receiving, decoding and playing media signals, also provide the user with a set of tools to help them decide their preferred content. The increasing offer of multimedia content made available through online platforms, such as Netflix or Amazon Prime, makes its selection process a demanding and tedious task for users and service providers. However, its particular relevance cannot be understated, since success of these platforms can often depend on a careful selection of content with proven success and guaranteed quality. Even though the usage of specialized selection techniques, applied to specific groups, is still the most preferable solution for the scenarios faced on multimedia systems, using users themselves as a mean to extract feedback is an alternative, interesting and cost-effective solution. For example, applying gaming paradigms to content selection and integrating these ideas on an overall solution can lead to an interactive experience where the user itself becomes a part of the solution, is rewarded by this initiative and improves the selection of multimedia content available on the system.

On the other hand, seamlessly integrating selection mechanisms on new or already existing solutions requires careful planning, with a particular focus on the interaction solutions used for this particular purpose. Traditional content selection systems typically rely only on physical controls, such as keyboard and mouse, to gather input information from the users. No additional data is gathered from other sources. Even though valuable information, such as typing speed or keys, can be used to provide emotional state information from regular users environments (Dalvand et. al. 2012), a complete picture of the content feedback from the user is not achieved with these traditional solutions.

On the other hand, Human Computer Interaction (HCI) enables interaction between users and computing devices which surround him, without any actual physical contact. HCI allows innovative and dynamic approaches to environments, since users are able to rely on pre-defined gestures to complete a set of tasks associated with a given tool. This kind of interaction was popularized on home gaming consoles, where low-cost hardware devices

were coupled with interactive gaming applications, achieving relative success. As a direct consequence of the use of HCI on modern devices, research on gestural interfaces was intensified. Correlating design approaches with successful interactive solutions led to significant contributions on the development of new interactive graphical user interfaces. However, the lack of established guidelines for gestural control, coupled with the misuse of established conventions, led to poorly designed Graphical User Interfaces (GUI) which were not adapted to interactive environments (Norman et. al. 2010).

On the technological front, the launch of Microsoft Kinect (Microsoft 2016) on November 2010 provided a new approach that was made available to researchers and general public. Combining a camera sensor, infrared projector and a special microchip used for 3D reconstruction, this hardware was capable of delivering depth maps and provide gesture and facial recognition to other applications. In the meantime, more advanced hardware solutions such as Intel RealSense (Intel 2016) have emerged, providing easy to use APIs that allow for easy integration on existing projects, while giving developers the means to improve existing tracking algorithms currently available on the hardware. In this work, a novel interactive solution for multimedia content selection is presented. Based on innovative forms of interaction such as hand gestures to attract users, this solution provides a gaming-like approach to tasks such as feedback extraction. Content categories, composed by sets of specific videos, are selectable by users. Users are able to approve/reject content and select a set of keywords which best represents that particular video. All videos will be presented within a specified time limit, during which the user will have to complete this task. When this process is completed, results for each video are presented and are able to be saved for further analysis. This solution was tested in a real-life scenario with a limited group of 10 people, the maximum amount achievable under the available circumstances. The positive feedback from 6 users from that overall user sample, along with minor suggestions from the remaining users, which rejected the solution, confirms the usability of this work and its applicability for this specific purpose. The tool is capable of being easily used for multimedia content selection, allowing users to complete this task without any major hiccups. This was the main milestone set for the development of this solution.

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Towards a collaborative system to check the reliability of information shared on social media applications

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Author Keywords. Social Media, Information, System, Auditability

Social media applications are able to generate great impact about a particular subject, regardless of the provenance of information. This kind of system provides first-hand data, but one pressing problem is to distinguish true information from misinformation, rumors and hoaxes. In many cases, user generated social media data can be biased, inaccurate, and subjective (Abbasi and Liu 2013). The idea is that social media users would be included in a collaborative process of content enhancement with the encouragement of information evaluation through the use of a system aimed at this type of action.

The existence of numerous social media applications encouraging the sharing of information combined with people's excitement on share their thoughts and activities can contribute to the spread of unreliable information. The publication and sharing of information without credibility in social media applications creates a misinformation cycle with irrelevant content and dangerous bias. Information published without any verification by users increasingly unconcerned about its validity, negatively contribute to destabilize even established truths. The growing number of people consuming information from social media applications contrasts with the discredit of the same people about the truthfulness of the spreaded content.

After literature review, we verified that suggested solutions to deal with the spread of unreliable information on the Internet is based on algorithms, disregarding characteristics of human behavior (Zhang and Leung 2015). Some works analyze ethical, semantic and epistemological dangers about false content in social media or how to build quality of information in this kind of systems (Hocevar, Flanagin, and Metzger 2014); however, in these studies we notice the concern and the need for information auditability circulating on the Internet, mostly on social media applications.

The purpose of this study is to analyze the behavior of users using a system that give to them the chance to audit the information they consume. Will the users be more cautious and will they reduce the spread of false content on social media applications since they can check information details before to share it and get help in a collaborative environment?

To check the validity of this approach we started to build a system prototype. In early steps we did a Heuristic Evaluation of the prototype analyzing usability and fulfillment of requirements (Pinheiro 2015). The evaluation showed that the guidelines for development of a system that allows the auditability of the information on social media applications and usability foundations have been well applied in the prototype.

The expected outcomes of this research will be evaluated through a case study. We in-tend to involve groups of people to deal with information picked up on their news feeds through the Application Programming Interfaces (APIs) of social medias and mix this content with hoaxes and fake information. The first group will use our system prototype with auditability features turned on and the other group will do it with features turned off (including the features proper of collaborative systems). An example among of others system prototype features is a mechanism that can retrieve external links shared on social media and cross these data with our database of blacklisted content sources crawled from fact check websites. The users will be alerted if occurs a match between our collected blacklisted content and current information in their news feed.

The prototype will be validated to check if the system features to support audit are useful in scenarios were users faces difficulties regarding information reliability. The results of the case study will come from the evaluation regarding the credibility (trustable or not trustable) of the information that both user groups cited above will remark in the news feed of prototype. The experience will help us to test the hypothesis that if we have a system with tools to audit information available in social media applications, then un-reliable information will be checked by users. The target audience of the case study are users who use social media applications as source of information. In our prototype we will only consider text entries and URLs retrieved from social media APIs, since we consider that evaluation of reliability of images and multimedia is a wider field of study. The social media applications chosen for the research are Facebook and Twitter, also as example of fact check websites in Portuguese language we have Boatos.org1 and Verdade Absoluta[2].

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Side-channel attacks on browser TLS

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Author Keywords. side-channel attack, website fingerprinting, web application, encryption, privacy, traffic

Extended Abstract

Nowadays, the usage of Internet is everywhere. In an era when access to the Internet is done on a regular basis, it is of great importance to guarantee that people are doing it in a secure way, specially when it involves the sharing of sensitive information, such as personal and financial data. The great majority of web applications usually work through the client-server model, where the client side accesses the server side through a browser. When this happens, there is information which is inevitably exposed on the network and it is impossible to fully guarantee its security and/or privacy. With regard to the development of the WWW, we are at a stage in which there are already a wide range of mechanisms that aim to ensure to the user the maximum possible privacy and/or security, as is the case of encryption and access control. However, there are still numerous techniques that allow the extraction of information for virtually every type of Internet activity.

One of these techniques are the side-channel attacks, in which the target is the way the system is implemented, rather than the specific, abstract details. Side-channel attacks are based on statistical analysis of side-channel information. This information is defined as information leaked from an encryption device, like the sound produced or the varying power consumption by the hardware during the encryption computation.

This work focuses on a type of side channel named web object size attacks. In this type of attack, the statistical analysis usually culminates in the making of a fingerprint for web applications. This means that every different characteristic or behavior of a web application translates to the way the packets generated by it are organized, which is such that it can be easily identified afterwards. Stored profiles can then be compared with captured data from the application the user is accessing (Shi and Biswas 2014).

Side channel attacks are something to be concerned about because they can be generated and implemented quickly and using widely available resources, which means that it is within the reach of virtually anyone with minimal knowledge about computers and net-working. Furthermore, the time one has to spend in order to obtain sensitive information from a side-channel attack can be rather small (Chen, Wang, Wang and Zhang 2010).

This study targets the behaviour of the web applications with the largest traffic in the world, according to the Alexa list of the top websites. We compare these applications and show how they can be distinguished based only on web object size.

Approach

Firstly, in order to extract the information present in the encrypted packets, we had to decrypt them. Using the SSL Session Key log file, which contains the symmetric session key used to encrypt the TLS traffic, it is possible to decrypt it. Then, while using a protocol analyzer like Wireshark, we acessed each web application three times so that traffic was generated.

After exporting the traffic data onto a csv file, namely the number of packets and their total size, we displayed it resorting to a graphical representation.

Results

This small experiment shows that when large amounts of information are allied to powerful and computationally developed tools, side-channel attacks can be successfully performed. Attackers will be able to identify almost instantaneously what the user is accessing. The left graph shows the fingerprint for each website. Each bubble represents a packet of a website, with a scale of their size in bytes, and each agglomerate of bubbles represents a website. The graph on the right maps the different websites on a cartesian coordinate system, according to the number of packets of the website (x-axis) and the total size of the packets, in bytes (y-axis). Having these two dimensions allows the identification of web applications.

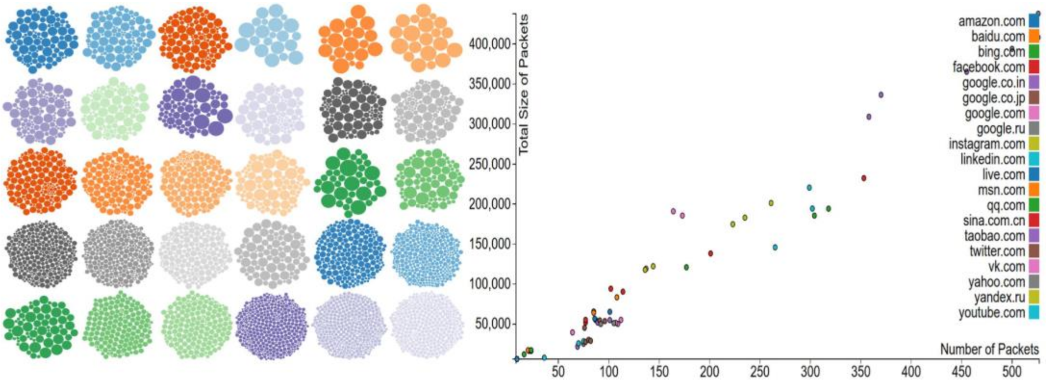

Figure 1: Graphs of HTTP Packets by web application

With the emergence of HTTP/2, it is no longer trivial to obtain information about sizes of web elements by capturing Internet traffic packets. This is because HTTP/2 enables pipelining, as well as the multiplexing of requests over a single TCP connection, which makes it difficult to identify web elements and their size (Morla 2016). This represents a challenge to fingerprinting attacks and something yet to be studied.

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Side Channel Attacks in 802.11

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Author Keywords. Wi-Fi, networks, privacy, Facebook, security

Extended abstract

Today, Wi-Fi networks are widely accessible. Data of all kinds is transferred, often containing private information that a user would not like to see in the hands of others. For this reason, encryption is applied in virtually every modern Wi-Fi network.

Conventional malicious attacks attempt to gather specific information from a user that may lead to a direct threat. An example would be that of an hacker trying to decrypt a user's communications in order to obtain their password, allowing them to perform potentially dangerous actions. For this type of attack, strong encryption can be effective as a counter-measure. However, encryption often does not hide everything. Through a variety of methods known as "side-channel attacks", it is possible to retrieve information regardless of encryption (Chen 2010). These attacks differ from their more conventional counterparts in that the attacker is outside the system and does not attempt to gain access to it. In fact, a side-channel attack relies on the characteristics of the transferred data, which can be observed from the outside, instead of its content.

Any normal activity a user may carry out on a website, such as accessing the website itself or scrolling through the main page leads to packet exchanges in particular patterns. An eventual attacker can identify the action with some level of certainty, based on their previous knowledge of the relationships between the patterns and the actions (Wang 2015). In order to carry out such an attack, the perpetrator must first prepare by passively collecting the exchange of packets between a device he is in control of and a network's access point, and use these to identify the unique pattern a certain action yields. The attacker then uses these previous statistical relationships to obtain his results.

In this paper, we put ourselves in the role of such an attacker and try to identify some actions within the Facebook website. By means of analyzing short (1-2 min) packet captures, we gather statistical data, which we can then use to ascertain what kind of activity a user is

performing within the website. For this paper, we decided to restrain ourselves to a limited number of actions.

Approach and methodology

In order to be able to detect which type of activity a user is performing on the Facebook website, we needed to have a statistical dataset, since the only way to be able to discern the activity is to compare it to what we know it looks like, in terms of several parameters. For this reason, we set up a network in FEUP's network laboratory in which we could do testing.

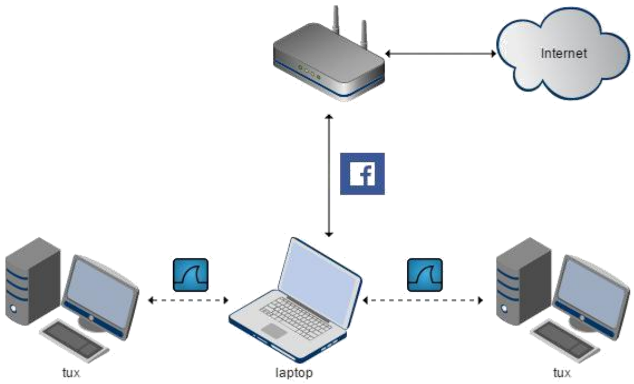

Figure 1: Network diagram

The main part of our controlled network is the laptop, which the victim is using to browse the website. In parallel, two desktop computers with a wireless network adapter configured in monitor mode perform eavesdropping on the communications between the laptop and the access point, also located in the laboratory. The apparent redundancy of using two computers is justified by the need to verify that the gathered data is the same independently of the device the attacker is using. The eavesdropping is accomplished using the Wireshark software. We then plotted and compared the shape of instantaneous bitrates for different actions on Facebook.

Results

After obtaining a solid dataset, we designated some features through which we could identify an action.

Action	Feature					
	Packet Count	Average Packet Size (bytes)	Burst duration (s)	Burst rate	Average Mbytes/s	Average Packets/s
Open Site	4502	867	4,31	3,17	0,26	300
Open Page	2362	918	1,91	2,2	0,22	236
Open Profile	1670	832	2,34	1,65	0,14	167
Scroll Feed	7228	880	1,17	3,54	0,06	71
Scroll Page	9508	952	3,98	3,99	0,18	190
Scroll Profile	2790	792	1,04	2,32	0,04	56

Figure 2: Values of the different parameters for each action

As we can see in figure 2, each action can be characterized by the values of the different parameters. Orange values show high variability, thus they have little statistical meaning. On the other hand, the remaining measures show low deviation from the presented average, which means they may be used to characterize the action. The high number of parameters considered makes for an easier differentiation between actions. Therefore, we concluded that it is possible to identify the actions performed by a user.

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Poster Session

Automatic Transcription of Vocalized Percussion

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Author Keywords. Beatbox transcription, music information retrieval, digital signal processing, assistive music technology.

Abstract

The primary goal of this work is to create an application that enables music producers to use their voice to create drum patterns when composing in music Digital Audio Workstations (DAWs). An easy-to-use and user-oriented system capable of automatically transcribing vocalizations of percussive sounds, called LVT, is presented. In order to achieve this goal, a Max for Live device was developed which follows the "segment-and-classify" methodology for drum transcription. LVT includes three modules: i) an onset detector to segment events in time; ii) a module that extracts relevant features from the music content; and iii) a machine-learning component that implements the k-nearest neighbors algorithm for classification of vocalized drum timbres.

In order to achieve the best possible results, two different approaches were tested. The first one uses a vocalized percussion dataset to train a generalized, user-independent transcription system, while in the second, a specific end-user trains the algorithm with their own vocalizations for each drum sound. A Max external that implements the sequential forward selection for choosing the features most relevant for their chosen sounds is proposed as well as a new annotated dataset of vocalized drum sounds.

Our ongoing evaluation shows that the user-trained system is able to very accurately identify the given vocalizations which outperforms the dataset-trained system as well as the built-in drum transcription system in Ableton Live.

Introduction

With changes in music culture, music production and how musicians work with their instruments have also changed. In other words, the ability to invent and reinvent the way to produce music is key to progress. Consequently, new proposals are necessary, such as designing new techniques for the composition of music. Within the genre of Electronic

Music the sequencing of drum patterns plays a critical role. The voice is an important and powerful instrument of rhythm production so it can be used to express or "perform" drum patterns in a very intuitive way. In order to leverage this concept within a computational system, we create a system that can help users (both expert musicians and amateur enthusiasts) input the rhythm patterns they have in mind to a sequencer via automatic transcription of vocalized percussion. Our proposed tool is beneficial both from the perspective of workflow optimization (by providing accurate real-time transcriptions), but also as means to encourage users to engage with technology in the pursuit of creative activities.

Methodology

A vocalized drum transcription software, able to be trained with the user vocalizations is proposed. Our system has been developed as a Max for Live project. Max for Live is a visual programming environment, based on Max 7 (www.cycling74.com), which allows users to build instruments and effects for use within the Ableton Live digital audio workstation.

To develop our system, a dataset of vocalized percussion was compiled. It was annotated using Sonic Visualizer, a free application for viewing and analyzing the contents of music audio files. The recordings, obtained from a set of 20 users with different backgrounds and levels of experience in beatboxing were saved and organized in an Ableton Live project file for compatibility and to facilitate the testing of existing drum transcription systems.

To approach the task of percussion transcription, we developed a generalized approach trained across the entire dataset, as well as user-specific version. Both systems use the high frequency content approach for onset detection, which is shown to be good method for identifying non-pitched percussive onsets. The module that follows the onset detection is feature extraction. A set of temporal and spectral features are extracted from the incoming audio signal when an onset is detected. The temporal features are the RMS value of the energy and the number of zero crossings, while the spectral features are the spectral centroid, spread, slope, decrease, rolloff, flatness, skewness, slope, flux and kurtosis; the Mel-frequency cepstrum coefficients and the bark frequency cepstral coefficients. For both the general and user-specific approaches, the sequential forward selection method was used to adapt the choice of features – which are then passed to a kNN classifier to make the final transcription. For the user-specific approach, the features are selected given a set of examples of each drum type. A screenshot of the interface for the user-specific version, LVT

is shown in Figure 1. It demonstrates the user-specific training stage – where a user inputs a set number of the drum timbres they intend to use, after which their vocalized percussion is transcribed and rendered as a MIDI file for subsequent synthesis.

Figure 1: User Interface of the LVT

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Underwater Power Transfer based on Resonance Magnetic Coupling

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Author Keywords. Wireless Power Transfer, Magnetic Coupling, Underwater Powering Systems, AUVs.

Extended Abstract

There has been an emergent request of autonomous underwater vehicles (AUVs) for several underwater applications, such as sea monitoring, fish tracking and identification, biological characterization of the sea bottom, amongst others Assaf 2013. These kind of routines usually require long-term operations, which in general are limited by the energy autonomy of the AUV.

Nowadays the process of recharging the batteries of the AUV still relies on a very time consuming task requiring human interaction. In general it is accomplished by removing the vehicle out of the water and charging it by the means of wet-mate connectors. Therefore, to extend the AUV autonomy, the battery charging process can substantially benefit from a wireless powering scheme if such a system is integrated within a underwater docking infrastructure. This can be achieved using a powerful transmitter system at the docking station, whereas the AUV comprises an efficient rectifier system and circuitry to regulate the charge rate of batteries. In this underwater scenario, each side should comprise a coupling coil, i.e. both the power transmitter (at the docking station) and the receiver (i.e. the AUV) need to be properly aligned so that their coils can build together a magnetic link between

them, which is used to transfer energy and recharge the batteries of the AUV.

There are several challenges in this approach. For instance, besides aligned, the coils should be preferably the closest possible to maximize the magnetic coupling during the charging process. However, to avoid changes in the electrical parameters along time, each coil requires a sealed housing built in accordance with the docking structure, which poses several physical restrictions also due to the geometries of the AUV.

This limits the minimum distance between the transmitter and receiver, with impact on the possible coupling coefficient ( ), which is typically in the order of 0.1 to 0.3 for distances of 3 to 5 centimeters between AUV and docking structures.

Such a reduced value of has its negative implications in terms of power transfer operation, and in the case of salt water, also the inherent conduction losses affect the system in terms of efficiency Kuipers 2014. The present work addresses the aforementioned challenges in powering an AUV. We re-port the study and implementation of a system developed for transmitting and receiving wireless power in sea water, focusing on the design of the coupling coils and the electronic system required.

The transmitter comprises an XMC-4400 microcontroller to synthesize pulse-width modulation (PWM) signals used to drive a half-bridge class-D power amplifier using gate drive bootstrapping. The system is powered by a battery bank with nominal voltage 13.2V (4.2Ah). Current-sensing is performed at the primary side to adapt system parameters. Both frequency tuning and PWM adjustment is possible using the XMC microcontroller. At the secondary a Schottky full-bridge is employed followed by a wide-range buck-boost to deliver a 18V regulated voltage. The system has been experienced in several forms, in particular with two different resonance topologies, the series-series (SS) and series-parallel (SP), and different coils (distinct number of turns, twisted, etc). We employed a simplified model of the system, implemented in a circuit simulator (Spectre from Cadence). Fig. 1 depicts the results for the power output and efficiency, demonstrating reasonable agreement between model and measurements.

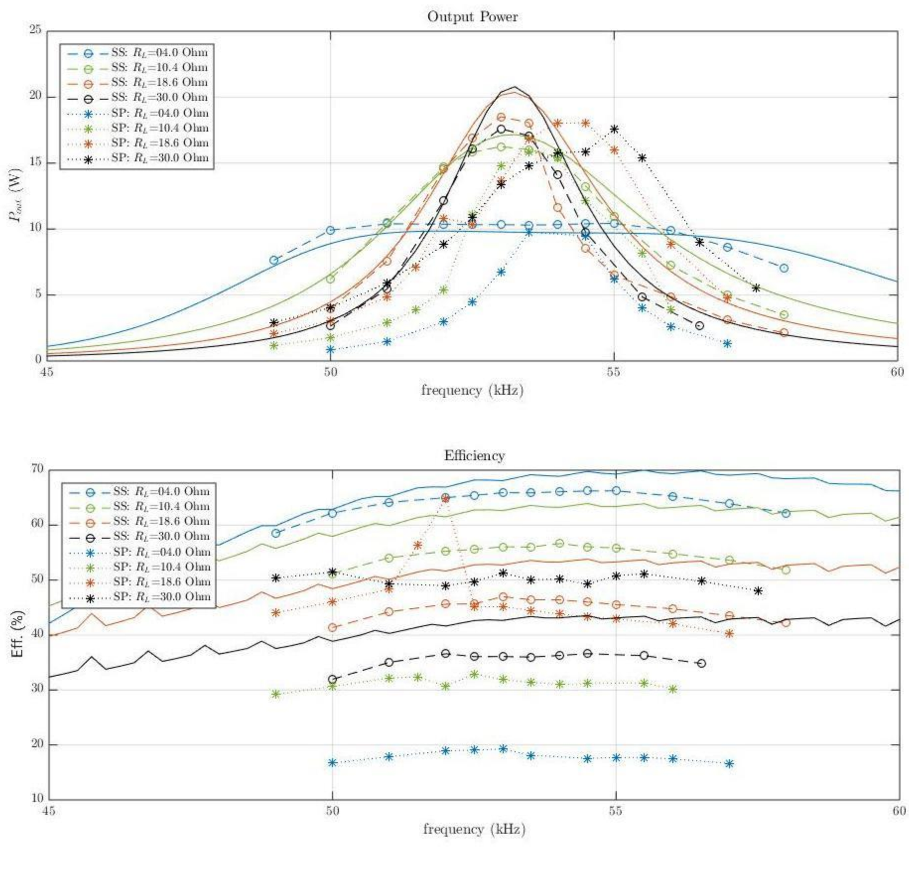

Figure 1: Output power (top plot) and efficiency (bottom plot) for simulation (solid lines) and measured data (dashed/symbols) in two configurations, i.e. series-series (SS) and series-paralel (SP) topologies.

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Awards

Best oral communication award for the Symposium on Electrical and Computers Engineering

• João Correia. Power Amplifier Modeling using Neural Networks for Predistortion;

• João Paulo Caetano Pereira. Musically-Informed Adaptive Audio Reverberation;

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