Advanced Techniques for the Characterization and Experimental Validation of Passive Inter-Modulation Effect (PIM) in Space Communications Systems

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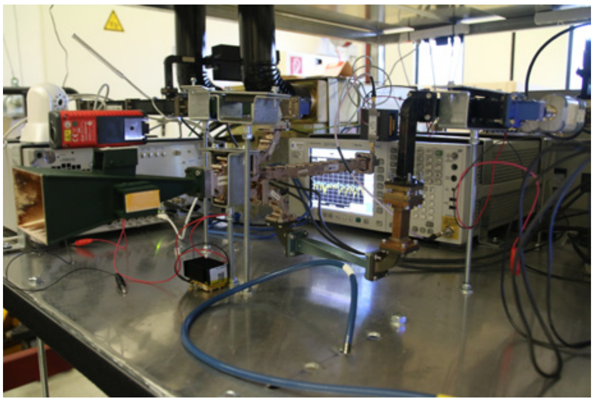
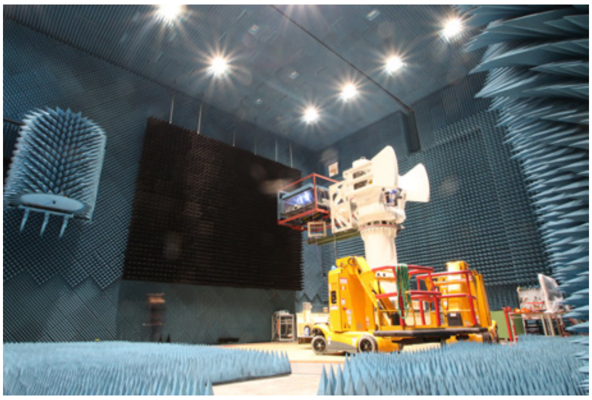
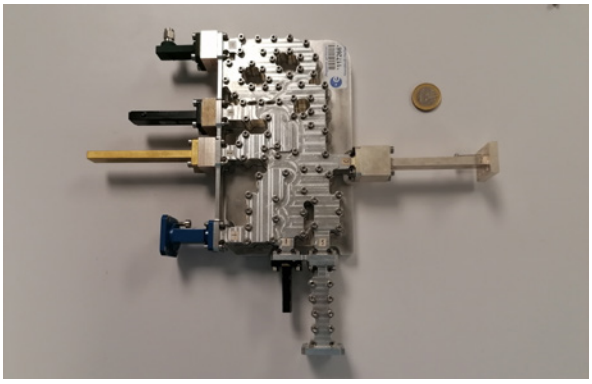
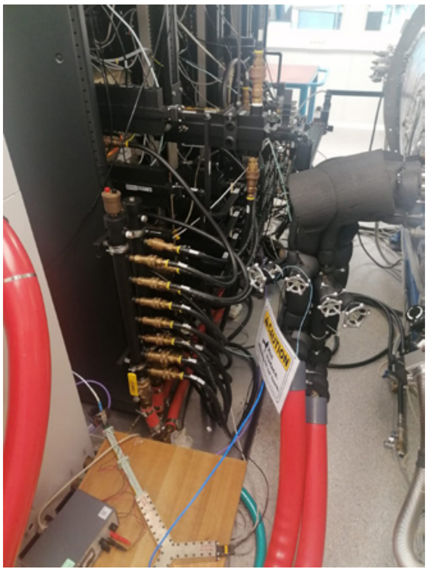
SUMMARY

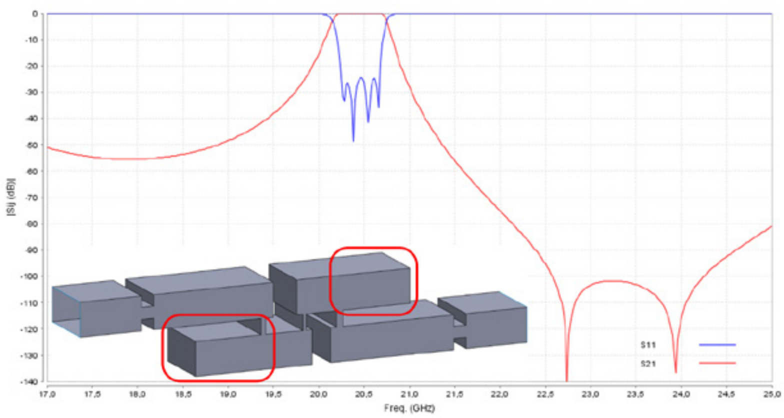
Modern satellite payloads operate in multicarrier scenarios, under a continuous demand for higher capacity links. This leads to an increase in the RF power levels, frequency of operation, and the number of transmitted channels, thus stimulating non-linear high-power effects, such as Multipactor, Corona, thermal issues and Passive Inter-Modulation (PIM). Among the above-mentioned phenomena, PIM is the less studied, or, at least, understood. This is due to its extreme non-linear nature and its close relation to workmanship, which make very difficult the development of models able to faithfully predict and explain PIM degradation. PIM terms, once ignited in the downlink, may interfere the weak signal to be detected in the uplink channel, thus threatening the payload throughput. Traditional PIM models are based on a two-carriers excitation. This is a simple and quite representative case, but has significant differences with the real multi- carrier scenario. This Ph.D. thesis work tries to diminish this gap by two novel contributions of relevance for real operation conditions. Firstly, the role of the carrier phases (neglected for two-carriers excitation) has been theoretically investigated. Secondly, a new model to account for the effect of non-contributing carriers for a given PIM term has been developed, which is based on a novel energy conservation assumption. The resulting models fit to experimental data. Due to the complexity of PIM modeling, PIM validation of RF components is conducted only by testing. The availability of low PIM test set-ups is therefore of great interest for the space industry. However, the design of low PIM test benches is challenging, as their intrinsic residual PIM has to be below the one requested to validate the test devices. For satellite hardware, the dynamic range between the RF power levels of the transmission carriers and the signal to be detected may be 185 dBc. During this Ph.D. thesis work, novel integrated test bed architectures in waveguide technology, both for conducted and radiated PIM scenarios, have been developed. These architectures consent a mitigation of the residual PIM of the test facility, being at the same time flexible, free from unwanted interactions and spurious resonances, and able to withstand considerable RF power levels for the transmission carriers. The key elements of these set-ups are the low PIM multiplexers, which may integrate two new families of waveguide filters able to provide a high number of transmission zeros, and therefore a high rejection, in the PIM reception channel. The test benches conceived for measuring conducted backward PIM, however, are normally unprotected from the PIM generated by the termination absorbing the high-power transmission carriers. To alleviate this situation, a new type of low PIM terminations in waveguide technology has been proposed and verified with PIM tests, showing a clear benefit in mitigating the residual PIM of the test facilities. Moreover, novel transitions able to improve the PIM performance of standard flanges have also been conceived. Finally, and with regard to radiated scenarios, a novel formulation able to convert payload PIM specifications to a practical PIM test is proposed. This formulation consents to link the power flux densities at the device under test (DUT) with the RF power levels measured by the test bench. Last, a large class of PIM measurements carried out with the novel test bed architectures have been reported. These measurements cover several frequency bands (C-, Ku-, K- and Ka) and different PIM scenarios, both conducted and radiated. The exceptional residual PIM noise floor of each test bed will be pointed out. In addition, PIM tests on an anechoic chamber facility, multi-layer insulation blankets (MLIs) and reflector mesh samples are presented, with interesting considerations about the geometry of the structure and the impact on the PIM performance of typical elements as sawing areas and rivets.

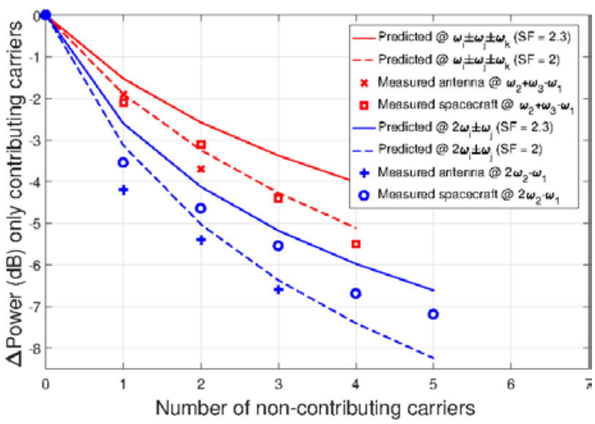


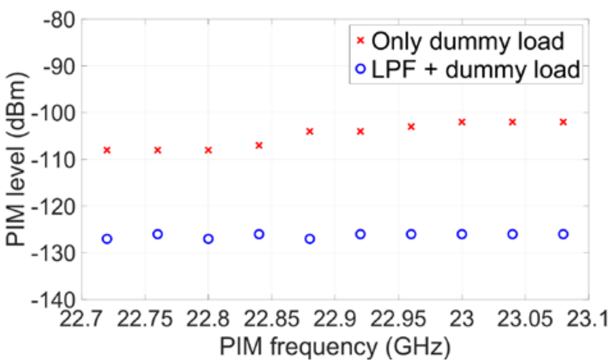







|Sij (dB)|
Freq. (GHz)
S11
S21


Predicted @ ω_i±ω_j±ω_k (SF = 2.3)
Predicted @ ω_i±ω_j±ω_k (SF = 2)
Measured antenna @ ω_2+ω_3-ω_1
Measured spacecraft @ ω_2+ω_3-ω_1
Predicted @ 2ω_i±ω_j (SF = 2.3)
Predicted @ 2ω_i±ω_j (SF = 2)
Measured antenna @ 2ω_2-ω_1
Measured spacecraft @ 2ω_2-ω_1
ΔPower (dB) only contributing carriers
Number of non-contributing carriers


Only dummy load
LPF + dummy load
PIM level (dBm)
PIM frequency (GHz)

Smart sound control in acoustic sensor networks: a perceptual perspective

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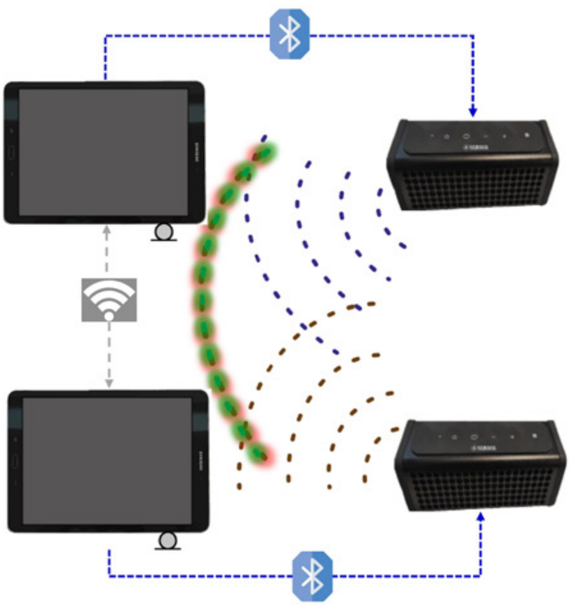
Supervisor: Dr. Gema Piñero Sipán and Dr. María de Diego Antón

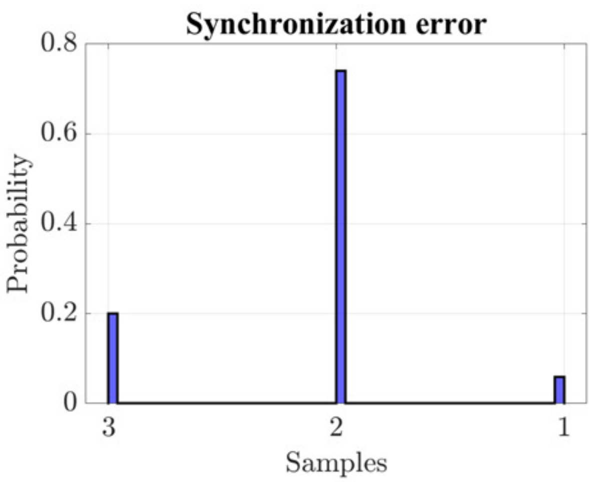
Defended on February 24th, 2022

SUMMARY

Audio systems have been extensively developed in recent years thanks to the increase of devices with high-performance processors capable of performing more efficient audio processing. In addition, the expansion of wireless communications has given the possibility of implementing networks in which devices can be placed in different locations without physical limitations, unlike wired networks. The combination of these technologies has led to the emergence of Acoustic Sensor Networks (ASN). An ASN is composed of nodes equipped with audio transducers, such as microphones or speakers. In the case of acoustic field monitoring, only acoustic sensors (or microphones) need to be incorporated into the ASN nodes. However, in the case of control applications, the nodes must interact with the acoustic field through loudspeakers.

The ASN can be implemented through low-cost devices, such as Raspberry Pi or commercial mobile devices, capable of managing multiple microphones and loudspeakers and offering good computational capacity. In addition, these devices can communicate through wireless connections, such as Wi-Fi or Bluetooth. This ASN design provides high processing power and flexibility due to the processors and the wireless communications offered by the current mobile devices. Therefore, in this dissertation, an ASN composed of commercial mobile devices connected to wireless speakers through a Bluetooth link is proposed. Additionally, the problem of synchronization between the devices in an ASN is one of the main challenges





to be addressed since the audio processing performance is very sensitive to the lack of synchronism. Therefore, a deep analysis of the synchronization problem between commercial devices connected to wireless speakers in an ASN is also carried out. In this regard, one of the main contributions is the analysis of the audio latency of mobile devices when the acoustic nodes in the ASN are comprised of mobile devices communicating with the corresponding loudspeakers through Bluetooth links. A second significant contribution of this dissertation is the implementation of a method to synchronize the different devices of an ASN, together with a study of its limitations. Finally, the proposed method has been introduced in order to implement personal sound zones (PSZ) applications. Therefore, the implementation and analysis of the performance of different audio applications over an ASN composed of commercial mobile devices and wireless speakers is also a significant contribution in the area of ASN

In cases where the acoustic environment negatively affects the perception of the audio signal emitted by the ASN loudspeakers, equalization techniques are used with the objective of enhancing the perception threshold of the audio signal. For this purpose, a smart equalization system is defined and implemented in this dissertation. In this regard, psychoacoustic algorithms are employed in order to implement a smart processing based on the human hearing system capable of adapting to changes in the environment, and thus increase the perception threshold of the audio signal dynamically. Therefore, another important contribution of this thesis focuses on the analysis of the spectral masking between two complex sounds. This analysis will allow to calculate the masking threshold of one sound over the other in a more accurate way than the currently used methods. This method is used to implement a perceptual equalization application that aims to improve the perception threshold of the audio signal in presence of ambient noise. To this end, this thesis proposes two different equalization algorithms: 1) pre-equalizing the audio signal so that it is perceived above the ambient noise masking threshold and 2) designing a perceptual control of ambient noise in active noise equalization (ANE) systems, so that the perceived ambient noise level is below the masking threshold of the audio signal. Therefore, the last contribution of this dissertation is the implementation of a perceptual equalization application with the two different embedded equalization algorithms and the analysis of their performance through the testbed carried out in the GTAC-iTEAM laboratory.

The wind orchestra as a means of application and development of current compositive techniques. A performative proposal

Author: José Miguel Fayos Jordán
Supervisor: Dr. Carlos Fontcuberta Llavata and Dr. Jorge Sastre Martínez

Defended on May 4, 2022

SUMMARY

The aesthetic revolution that the twentieth century has seen, where a multitude of proposals arose, reached the different instrumental groups in a disparate way. While symphonic and chamber music lived a rebirth sponsored by new compositional techniques, the wind orchestra was stuck in an apparent context of tonality and modality that, far from evolving at the same time as the other formations, has experienced an apparent retreat towards postures more conservative aesthetics.

The present work arises from the need to evolve the current repertoire for wind orchestra, deepening its timbral possibilities and developing a repertoire that addresses composition for wind orchestra, from a current perspective that develops some of the main aesthetic currents of the avant-garde musical of the twentieth and twenty-first centuries, such as spectralism, algorithmic composition or the use of techniques derived from the scientific-mathematical field. Next, a detailed analysis of the compositional processes addressed in each of the works under study is elaborated, without losing the perspective of the fundamental objective; the composition of two works that develop the sound potential of the orchestral wind and percussion formation within a contemporary style.

Previously, some of the most important contributions to the genre made by different composers are exposed, which have meant a real and significant connection of the new compositional proposals of the 20th and 21st centuries with the formation of winds in their orchestral or large ensemble

Variables:	A, B, C, D
Axioms:	(A←B), (B←AB), (BB←CAA), (C←ADB), (D←ABC)
n = 0 :	A
n = 1 :	B
n = 2 :	AB
n = 3 :	BAB
n = 4 :	ABBAB
n = 5 :	BCAABAB
n = 6 :	AADBBBABBAB
n = 7 :	BBABCCAAABBCAABAB
n = 8 :	CAABABADBADBBBBCAAADBBBABBAB
n = 9 :	ADBBBABBABBABCABBABCCAACAA...

concretion. Especially noteworthy are those proposals by authors closely related to the wind repertoire, especially in its band aspect. At the same time, other works by significant authors within the orchestral and chamber repertoire are referenced, who have made sporadic contributions to the wind orchestra / ensemble. Contributions that for their originality or style framed in the aesthetic compositional currents of the XX and XXI, imply an outstanding fact for the wind and percussion genre.

New Wide-Band Capacitive Filter Structures in Rectangular Waveguide Technology With Enhanced Out-of-Band Response

Author: Joaquin Francisco Valencia Sullca
Supervisor: Dr. Vicente E. Boria Esbert, Dr. Santiago Cogollos Borrás and Dr. Marco Guglielmi

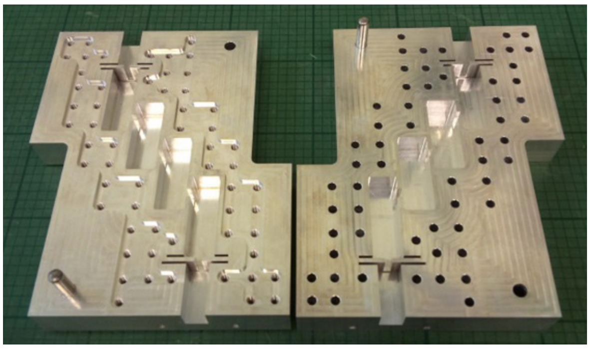
Defended on December, 16, 2021

SUMMARY

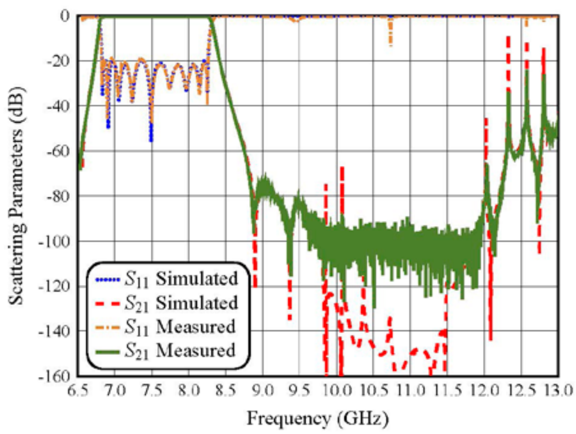
The main objective of this doctoral thesis is the development of new capacitive filtering structures in rectangular waveguide that are able to provide wide bandwidths in the passband and improve, at the same time, the out-of-band response. These new guided structures have been developed in order to offer new technological solutions for high-frequency microwave filters, with a variety of different transfer functions, addressing specifically the needs of future telecommunication systems for both ground and space applications.

In this context, therefore, we discuss in this document the study, design and manufacture of several types of microwave filter in rectangular waveguide that show a significant improvement with respect to the state-of-the-art. The new solutions that we propose are obtained by introducing simple modifications in the structure of classic microwave filters. Several techniques based on hybrid irises, stepped impedance resonators, staircase configuration and E-plane T-junctions with shorted stubs or manifold connections, are successfully used in order to meet the very demanding specifications of future systems for both ground and space applications. Furthermore, an additional current challenge faced by all designers of microwave components is the need to reduce both their physical size and mass (or weight). To address this issue, we discuss in this document the use of resonant apertures (RAs) in rectangular waveguide, introducing a new family of filters which can be used to implement complex single and dual-band transfer functions with significant size and mass reduction.

In this doctoral thesis, each subject is discussed in detail including the basic theoretical formulations, design procedures, the results of full-wave electromagnetic simulations, manufacturing considerations, and the measured performance of a number of prototypes. Excellent agreement is found in all cases between measurement and simulations, thereby fully validating both the novel structures discussed and their design procedures.



Manufactured extended bandpass RA prototype in aluminum (no silver plating)..



Measurement of the out-of-band performance of the extended bandpass RA filter compared with the EM simulation (CST).

THESIS SUMMARY

Evaluation of QoE in a DASH-based 3D video adaptive streaming system

Author: Paola Guzmán Castillo
Supervisor: Dr. Juan Carlos Guerri Cebollada and Dr. Pau Arce Vila

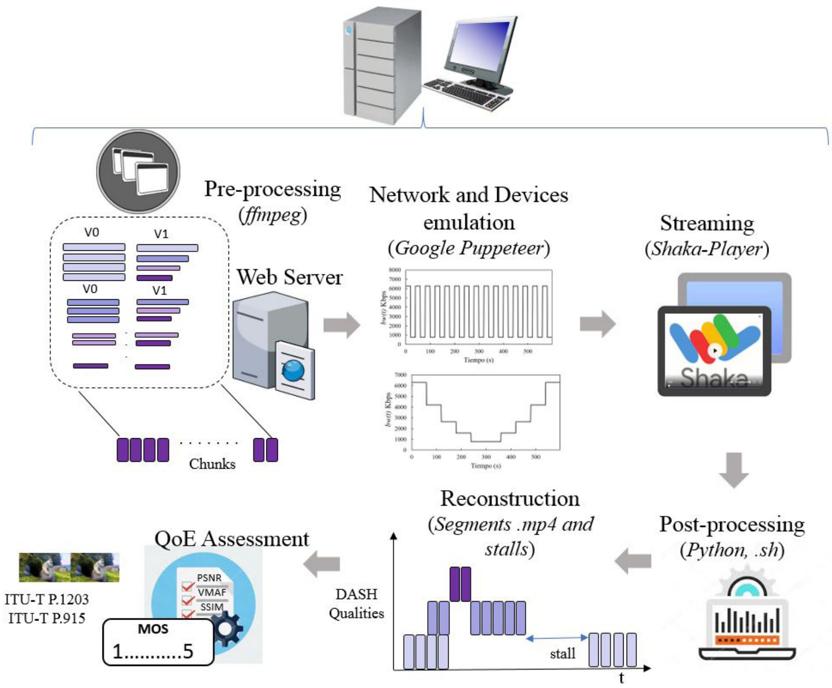
Defended on July 21, 2022

SUMMARY

The distribution of multimedia content, and in particular video streaming, currently dominates global Internet traffic and will become even more important in the future. Thousands of titles are added monthly to major service providers such as Netflix, YouTube and Amazon. In addition to the consumption of high-definition content becoming the main trend, an increase in the consumption of 3D content can be observed again. This fact has caused that issues related to content production, encoding, transmission, Quality of Service (QoS) and Quality of Experience (QoE) perceived by users of 3D video distribution systems became a research topic with numerous contributions in recent years.

This thesis addresses the problem of providing 3D video streaming services under variable bandwidth network conditions. In this sense, it presents the results of the evaluation of the QoE perceived by the users of 3D video systems, analyzing mainly the impact of the effects introduced in two of the elements of the 3D video processing chain: the encoding stage and the transmission process.

To analyze the effects of the encoding process on the quality of 3D video, the first stage deals with the objective and subjective evaluation of video quality, comparing the performance of different encoding standards and methods, in order to



identify those that achieve the best ratio between quality, bit rate and encoding time. Also, in the context of transmission in a simulcast environment, the advantages of using asymmetric coding for 3D video transmission is evaluated as an alternative for bandwidth reduction while maintaining overall quality.

Secondly, for the study of the impact and performance of the transmission process, the work has been carried out on the basis of an adaptive dynamic over HTTP (DASH) transmission system in the context of both 2D and 3D video transmission, using different bandwidth variation scenarios. The aim has been to develop a framework for the evaluation of QoE in 3D adaptive video streaming scenarios, which allows analyzing the impact on the user's QoE against different bandwidth variation patterns, as well as the performance of the adaptation algorithm under these scenarios. The work focuses on identifying the impact on the user's Quality of Experience in aspects such as: frequency, type, range and temporal location of bandwidth variation events.

The proposed system allows to perform performance measurements in an automated and systematic way for the evaluation of DASH systems in the 2D and 3D video distribution service. The tool Puppeteer, the Node.js library developed by Google, has been used, which provides a high-level API to automate actions in the Chrome Devtools protocol, such as starting playback, causing bandwidth changes and saving the results of the quality change processes, timestamps, stops, etc. From this data, a further processing is performed that allows the reconstruction of the displayed video, as well as the extraction of quality metrics and the evaluation of the QoE of the users using the ITU-T P.1203 recommendation.

Filter Optimization for Personal Sound Zones Systems

Author: Vicent Molés Cases
Supervisor: Dr. Gema Piñero Sipán and Dr. Alberto González Salvador

Defended on July 1st 2022

SUMMARY

Personal Sound Zones (PSZ) systems deliver different sounds to a number of listeners sharing an acoustic space through the use of loudspeakers together with signal processing techniques. These systems have attracted a lot of attention in recent years because of the wide range of applications that would benefit from the generation of individual listening zones, e.g., domestic or automotive audio applications. A key aspect of PSZ systems, at least for low and mid frequencies, is the optimization of the filters used to process the sound signals. Different algorithms have been proposed in the literature for computing those filters, each exhibiting some advantages and disadvantages. In this work, the state-of-the-art algorithms for PSZ systems are reviewed, and their performance in a reverberant environment is evaluated. Aspects such as the acoustic isolation between zones, the reproduction error, the energy of the filters, and the delay of the system are considered in the evaluations. Furthermore, computationally efficient strategies to obtain the filters are studied, and their computational complexity is compared too. The performance and computational evaluations reveal the main limitations of the state-of-the-art algorithms. In particular, the existing solutions can not offer low computational complexity and at the same time good performance for short system delays. Thus, a novel algorithm based on subband filtering that mitigates these limitations is proposed for PSZ systems. In addition, the proposed algorithm offers more versatility than the existing algorithms, since different system configurations, such as different filter lengths or sets of loudspeakers, can be used in each subband. The proposed algorithm is experimentally evaluated and tested in a reverberant environment, and its efficacy to mitigate the limitations of the existing solutions is demonstrated. Finally, the effect of the target responses in the optimization is discussed, and a novel approach that is based on windowing the target responses is proposed. The proposed approach is experimentally evaluated in two rooms with different reverberation levels. The evaluation results reveal that an appropriate windowing of the target responses can reduce the interference level between zones.