Título: SIGNAL PROCESSING ALGORITHMS FOR DIGITAL HEARING AIDS

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Resumen: Hearing loss is a problem that severely affects the speech communication and disqualify most hearing-impaired people from holding a normal life. Although the vast majority of hearing loss cases could be corrected by using hearing aids, however, only a scarce of hearing-impaired people who could be benefited from hearing aids purchase one.

This irregular use of hearing aids arises from the existence of a problem that, to date, has not been solved effectively and comfortably: the automatic adaptation of the hearing aid to the changing acoustic environment that surrounds its user. There are two approaches aiming to comply with it. On the one hand, the ¿manual¿ approach, in which the user has to identify the acoustic situation and choose the adequate amplification program has been found to be very uncomfortable. The second approach requires to include an automatic program selection within the hearing aid. This latter approach is deemed very useful by most hearing aid users, even if its performance is not completely perfect.

Although the necessity of the aforementioned sound classification system seems to be clear, its implementation is a very difficult matter. The development of an automatic sound classification system in a digital hearing aid is a challenging goal because of the inherent limitations of the Digital Signal Processor (DSP) the hearing aid is based on. The underlying reason is that most digital hearing aids have very strong constraints in terms of computational capacity, memory and battery, which seriously limit the implementation of advanced algorithms in them. With this in mind, this thesis focuses on the design and implementation of a prototype for a digital hearing aid able to automatically classify the acoustic environments hearing aid users daily face on and select the amplification program that is best adapted to such environment aiming at enhancing the speech intelligibility perceived by the user.

The most important contribution of this thesis is the implementation of a prototype for a digital hearing aid that automatically classifies the acoustic environment surrounding its user and selects the most appropriate amplification program for such environment, aiming at enhancing the sound quality perceived by the user. The battery life of this hearing aid is 140 hours, which has been found to be very similar to that of hearing aids in the market, and what is of key importance, there is still about 30% of the DSP resources available for implementing other algorithms.