THE ROAD OF AN ACOUSTIC ECHO CONTROLLER FOR MOBILE TELEPHONY FROM PRODUCT DEFINITION TILL PRODUCTION

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ABSTRACT

This paper reviews under an industrial perspective the road followed by an acoustic echo controller (AEC) from product definition till final production. Particularly, we discuss the most significant aspects of the AEC development process, *i.e.* technical specifications, algorithm development and implementation, AEC parameters tuning, measurements and validation.

1. INTRODUCTION

The development and the implementation of an acoustic echo controller (AEC) for mobile telephony requires a careful methodology, necessary for guaranteeing standards compliance and end-customer satisfaction. Several tests and verifications have to be applied at any stage of the development process. Some of these tests derive from international standardization institutes but most of them rely on the own experience of the DSP company or the mobile phone manufacturer which implements the acoustic echo controller. This paper reviews under an industrial perspective the road followed by an acoustic echo controller from product definition till final production.

The acoustic interfaces of mobile telephone terminals may be divided in five fundamental classes [2, 3]: handset, headset, vehicle mounted hands-free, desk-top operated hands-free and hand-held hands-free. Different requirements and technical specifications are imposed to the mobile phone manufacturer for each of these classes. While headset equipments typically do not require any form of echo control, AECs become mandatory in all kind of hands-free interfaces. It is a common assumption that handset equipments do not need any echo control. In reality, the reduced size of modern handset, the acoustic coupling between earpiece and microphone, the seismic coupling through the mechanical part of the terminal and the electrical coupling due to crosstalk create a significant echo feedback between the receiving and the transmitting audio port of the mobile terminal. In these conditions, the adoption of some echo control technique is almost always mandatory. The problem is also exacerbated from the nonlinearities that may be present in the echo path [15, 17].

Depending on the product being developed, different implementation solutions are viable. The mechanical and electrical design of the product and the electro-acoustic components selection (*i.e.* the loudspeaker and the microphone selection) assume an important role in echo control scenario. By their own, design and component selection are not able to avoid the problem of acoustic echo; nevertheless, they can greatly facilitate the successive AEC development. We next consider the development of acoustic echo controllers for full-duplex communications, which, in their essence, are constituted by an adaptive echo canceller followed by some nonlinear processor for residual echo removal [16]. For some application the acoustic echo controller may be implemented in the central processing unit of the mobile phone (typically a micro-controller, a DSP or a combination of the two) while in other applications the adoption of a separate processing device (a DSP or an ASIC circuit) becomes necessary. The adopted implementation imposes constraints on the algorithm choice in terms of power consumption, computational complexity, memory requirements or silicon area. Furthermore, the product design, the target cost and time-to-market determine other severe constraints on the echo controller development.

In the next sections we comment the most significant industrial aspects of the AEC development process, *i.e.* technical specifications, algorithm development and implementation, AEC parameters tuning for the particular product, measurements and validation. The development procedure here presented is only one example of the many possible approaches. Methodologies surely change in any company depending on the background experience of the developers, on the available technology and instrumentation and on the particular product being developed. Nevertheless, we think

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that many of the discussed aspects are part of a common trend dictated by good-sense and by the need of ensuring quality.

In what follows, we refer to the technical specifications and the measurement recommendations provided by European Telecommunication Standard Institute (ETSI) for narrow-band telephony (*i.e.* 8 kHz sampling frequency) for the GSM and UMTS telecommunication system. Some comments on applicable ITU-T recommendations are also provided.

2. TECHNICAL SPECIFICATIONS

The declared objective of ETSI technical specifications for mobile telephony is to reach a quality as close as possible to ITU-T standards for Public Switched Telephone Network (PSTN). In order to overcome the problem of the acoustic echo a 46dB weighted Terminal Coupling Loss (TLCw) [7] should be achieved by the terminal [1, 2, 4, 6]. Indeed, according to the ITU-T Recommendation G.131 [8] the 46dB value of TCLw provides an adequate echo protection for calls with a delay up to 300ms, which is not uncommon in second and third generations mobile telephone networks. The TLCw is defined in accordance to the ITU-T Recommendation G.122 as "the integral of the power transfer characteristic (attenuation) $A(f)$ weighted by a negative slope of 3 dB/octave starting at 300 Hz, extending to 3400 Hz" [7]. While the 46dB TLCw is mandatory for handset and headset user equipment [1, 2], this value of echo attenuation can be hardly met with hands-free terminals and, most likely, it would require a undesirable strong action of the nonlinear processor of the echo controller. Moreover, the hands-free devices typically operate with more severe background noise conditions than handset and head-set terminals and background noise can efficiently mask a residual echo coming from user equipment [18]. For these reasons, some milder specifications have been imposed for hands-free devices. Particularly, the ETSI technical specifications [1, 2] impose for all hands-free devices a TCLw of "40dB at the nominal setting of volume control in quiet background and (of) 33dB at the maximum user selectable volume control setting." Technical specifications [1, 4] also allow an additional 39 ms delay "for additional processing for hands-free". It has to be remarked that this additional delay increase the one way delay budget of the 40% [18] and therefore it increases the listener sensibility towards echo. Moreover, this delay can not be applied to handset acoustic interfaces. Therefore, it is a good practice to keep the additional delay as low as possible. Another indirect specification for the echo controller comes from test set-up in [1] and [3] where it can be evinced that the convergence time of the echo canceller with a male or female artificial far-end voice [10] should be less than 10 s. This convergence time value appears absolutely unacceptable for guaranteeing a pleasant quality of end-to-end communication. A more reasonable value of the convergence time is given in the ITU-T Recommendation G.167 [9] where a 20dB TLCw is required after 1 second of artificial voice. ITU-T recommendations G.167 [9] and P.340 [13] provide several specifications for acoustic echo controller also in terms of performance under double-talk and echo-path chance conditions. Despite ETSI specifications does not require adherence to these recommendations, compliance to their requirements is always suggested.

3. ALGORITHM DEVELOPMENT AND IMPLEMENTATION

This is perhaps the most creative part of all the development process. Many suitable families of algorithms can be applied (see [16, 19] and references therein). Each of these algorithms has its own advantages and its own limitations. Since there is no "universal" solution, algorithm selection basically relies on the experience of the developer. He has not only to design the key components of the acoustic echo controller (*i.e.* the acoustic echo canceller and the nonlinear post-processor) but he has to specify and develop all those "auxiliary" functions (*e.g.* audio filters, level adjustment systems, background noise estimators, signal level estimators, echo-path change detectors, doubletalk detectors, etc.) that are fundamental for a reliable operation of the equipment [19].

The algorithm design must reflect the particular application of the echo controller. For this reason, an important tool for the AEC development is the availability of a database of acoustic echo signals. This database must be recorded from an acoustic interface as close as possible to that of the final product. Many times, the database is constituted by hours of recordings. The objective of the database is twofold. From one side it is a reference for the development of the AEC algorithm and for setting the values of the AEC parameters. On the other side the AEC signals are employed at any stage of the development process for testing the algorithm and its implementation by means of objective measurements or by third-party listening subjective evaluations [14].

A first screening of candidate algorithms can be obtained by simulations. In this stage of the development, objective measurements are typically applied in order to select a limited number of candidates, which appear the most promising for the particular application. Full subjective tests are then conducted on winning algorithms by "expert" listeners [14] in order to assess a pleasant audio quality. Candidate algorithms that fulfill specifications and guarantee listening comfort are then ported to the computational precision of the target application. Again the algorithm is simulated and tested with both objective and subjective evaluations in order to assess performance preservation. The last stage of the development process consists in the implementation on a target device, *e.g.* a DSP or an ASIC circuit. The AEC is emulated and it is newly tested in order to assess, again, specification compliance and pleasant listening quality. This tedious repetition of test procedures is necessary in order to guarantee error avoidance and end-product quality.

4. AEC TUNING

The integration of the acoustic echo controller with the final product almost always requires an accurate tuning of the AEC parameters in order to optimize performances for the particular application. Indeed, no simulated environment can fully reflect the constellation of environmental conditions that characterizes the normal use of a mobile terminal. Moreover, many times the mobile terminal implements some general purpose echo control algorithm that has to be necessarily adapted to the particular application for obtaining reasonable performances.

Parameters tuning is typically performed with a trial and error procedure by means of an extensive number of subjective tests. Conversational and double-talk subjective tests, performed by experienced subjects [14], are continuously repeated at any parameter modification. Parameters interdependency and algorithm complexity can stress the difficulty of this task. Moreover, the technicians performing parameter tuning many times do not have any experience of the particular AEC implemented. For example, this is often the case with the AEC implemented in the central processing unit of cellular phones: while the AEC algorithm is developed and implemented by the silicon vendor, product integration and parameters tuning is performed by the mobile phone manufacturer.

Good-sense rules for facilitating this task dictate during algorithm development the adoption of physically meaningful parameters, the avoidance of parameters interdependency and the choice of a limited number of modifiable parameters.

5. MEASUREMENTS

Measurements are objective methods for the assessment of the quality of a product. They are aimed at deriving meaningful quantities characterizing some properties of the observed object and at guaranteeing that the terminal, in any condition, will not damage any other equipment operating in the same telecommunication network. Measurements should be repeatable and reproducible. It is evident that these requirements contrast with the intrinsic variability of the acoustic echo, which is influenced by all environmental conditions. As a matter of fact, ETSI has standardized only two test that assess the quality of the acoustic echo control: the acoustic coupling loss test and stability margin test.

The acoustic coupling loss test [3, 5] is aimed at measuring the TLCw of the mobile terminal. Test condition should be as close as possible to real operating conditions. Hand-free terminal are setup in a room where they are intended to be used, *e.g.* for a vehicle-mounted hands-free the equipment should be tested in a vehicle or a vehicle simulator. Handset equipments are mounted on a HATS (head and torso simulator) [11], on a LRGP (Loudness Rating Guarding Position) [12] or are left in free air. A training sequence constituted by 10 [5] to 20 seconds [3] of male and female artificial voice [10] is applied to the terminal in order to allow the echo canceller adaptation. Then the TCLw is computed according to ITU-T Recommendation G.122 [7] by employing a test signal that is either an artificial voice, a logarithmically spaced multi-sine or a pseudo-noise sequence.

The stability margin [5] is a measure of the gain that would have to be inserted between the go and return paths of the reference speech coder for oscillation to occur. A stability margin of at least 6 dB is required. The test procedure considers for hands-free devices the normal operating condition, while handset devices are placed on a hard plane surface with the transducers facing the surface. The mobile equipment is operated and the transmit path signal is looped to the receive path with a 6dB amplification. By injecting an impulsive noise or a pseudo-random noise in the system no audible oscillation shall be detected. Interestingly, many times, under the particular conditions of the stability margin test, the design of the handset receiver or of the hands-free equipment creates an undesirable acoustic coupling between the loudspeaker and the microphone. A suitable acoustic echo controller is then mandatory in order to pass the stability margin test.

It is clear that the two tests we have just described are unable to characterize the performances of an acoustic echo canceller, nor they are intended to do that. Many other measurements are standardized by the ITU-T Recommendation G.167 [9]. Unfortunately, most of these measurements are not easily applicable to the mobile phone end-product. In practice, mobile phone manufactures integrate the ETSI tests with their own proprietary test procedures. These test procedures come from a long legacy of products, trials and errors and they concentrate only on a subset of the AEC parameters defined in [9]. Test pass or fail is often decided on the basis of comparisons with some reference phones that are representative of the state of art of mobile telephony.

6. VALIDATION

No measurement can replace the value of subjective tests performed in real operating conditions. The purpose of validation is to assess the overall quality of the mobile terminal as perceived by the end-customer and validation tests are typically repeated for every network provider. The AEC validation involves subjective conversational and double-talk tests performed both by "experienced" and "untrained" subjects [14]. Different guidelines for hands-free subjective tests may be found in ITU-T Recommendation P.832 [14]. The target product and other reference phones are tested on-field in different environmental conditions that try to replicate the end-customer normal operations. Subjects performing these tests provide their rating of the quality of the overall communication. A pass or fail decision is again based on comparisons with the performances of reference phones.

Interestingly, according to the subjective tests, the problems that impair the most the conversation are: audible speech clipping or distortions during double-talk due to the effect of the nonlinear processor; modulation of the background noise caused by automatic gain controllers or by nonlinear processors and, eventually, the disturbance caused by echoes, particularly the echoes due to the initial convergence, to echo-path change variations or to the cancellation residual.

7. CONCLUSION

This paper provides a synthetic description of the procedures necessary to take an acoustic echo controller for mobile telephony from algorithm till final production. Technical and practical specifications, algorithm development and implementation, AEC parameter tuning, measurements and validation have been discussed. Only a few guidelines are provided for acoustic echo controllers by international standardization institutes. Therefore, the development process rely heavily on the experience, the technology and the instrumentation of the DSP company or the mobile phone manufacturer that takes care of the development, the implementation or the integration of the acoustic echo controller.

8. REFERENCES

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