

Extended summary

Advanced Applications for

Multichannel Teleconferencing Audio Systems

Curriculum: Ingegneria Elettronica, Informatica e delle Telecomunicazioni

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Abstract. This thesis deals with advanced digital signal processing techniques applied to multichannel teleconferencing audio systems, focusing on their benefits and drawbacks. A teleconferencing system should provide a realistic representation of visual and sound fields, allowing a natural communication among participants anywhere in the world as they were all in the same room. In this context, the work faces several issues mainly concerning on multichannel acoustic echo cancellation. Furthermore, immersive audio reproduction and listening environment equalization are taken into consideration with the aim of improving the performance of conventional teleconferencing systems. All the novel contributions given in the aforementioned scenario are discussed in this thesis. As regards multichannel acoustic echo cancellation, new solutions for channels decorrelation are described [1][2][3][4][5]: they are especially conceived for stereophonic acoustic echo cancellation and based on the "missingfundamental" phenomenon. Moreover, a novel variable step-size approach for improving the convergence speed of adaptive algorithms applied to a stereophonic acoustic echo canceller is discussed [6]. Afterwards, in the field of immersive audio, efficient solutions for real time implementation of Wave Field Synthesis (WFS) [7][8][9] and two possible phase approximations of the driving functions for further lowering the computational complexity are given [10].



Moreover, a WFS-based digital pointing of line arrays is discussed centered on the idea of reproducing a virtual source through WFS formulas [8][11]. With the aim of applying WFS technique to real scenarios, a WOLA-based approach for Wave Domain Adaptive Filtering is discussed [12]. As regards listening environment equalization, the analysis of a novel multipoint magnitude equalization is presented [13][14]. Then, a new solution for phase equalization is introduced in order to improve the perceived audio quality [15]. Eventually, a minor contribution on an ESPRIT-based approach for bass enhancement is presented based on a high resolution spectral estimation technique and a subspace fundamentals tracker [16].

Keywords. Multimedia teleconferencing systems, Multichannel acoustic echo cancellation, Immersive audio reproduction, Audio equalization, Bass enhancement.

1 Problem statement and objectives

1.1 Multichannel teleconferencing systems: an overview

Nowadays, there is a large interest towards multimedia teleconferencing systems because of the increasing requirement for efficient communications and the development of advanced digital signal processing techniques. As a matter of fact, multichannel systems have become essential since listeners need spatial information to identify the speaker position. In this scenario, several issues have been faced mainly concerning on multichannel acoustic echo cancellation. Afterwards, immersive audio reproduction and listening environment equalization have been studied for improving the performance of conventional systems.

1.2 Multichannel acoustic echo cancellation

Acoustic echo cancellers are exploited in teleconferencing audio systems in order to reduce echoes due to coupling between the loudspeaker and the microphone [17]. In the presence of more than one participant, multichannel systems have to be taken into consideration for speaker localization. By the use of a two-channel system, it is already possible to obtain more realistic performance than the monochannel case. Unfortunately, besides the fact that more adaptive filters are needed, the linear relationship existing between the two channels generated from the same source brings to the "nonuniqueness problem". Indeed, the covariance matrix is very ill-conditioned and the solution depends on the speaker position in the transmission room which is not stationary, causing possible convergence problems [17]. Therefore, a method to reduce interchannel coherence must be introduced in order to obtain good echo cancellation performance. In the literature, several techniques have been proposed in order to weaken this relationship, divided into two main categories: methods based on direct alteration of the stereo signals (e.g., nonlinear functions), such as those proposed in [17][18][19][20][21][22][23][24][25][26][27], and other techniques based on the introduction of an external signal to both channels, mainly exploiting psychoacoustic phenomena (e.g., masked noise), such as those proposed in [28][29][30]. Moreover, this relation affects also the choice of the adaptive filtering algorithm [31]. Thus, algorithms that do not take into consideration the cross-correlation among channels could converge very slowly. As a matter of fact, adopting straightforward extensions of single channel algorithms is not the best solution. In addition, to outperform the difficult choice of the step-size in the adaptation, several variable step-size-based approaches have been proposed in the literature: they aim to solve the trade-off between faster convergence and lower steady-state performance, as discussed in [32][33][34][35][36][37][38].

In this work, novel contributions within the stereo acoustic echo cancellation have been described, mainly related to solutions to the nonuniqueness problem; furthermore, a novel approach for improving the convergence speed of adaptive filtering algorithms has been presented.



1.3 Improvement of conventional teleconferencing systems

The increasing interest towards multimedia teleconferencing systems contextually with the spread of digital signal processing techniques brought to the investigation of audio rendering algorithms in order to improve the perceived audio quality and to make the listening environment more pleasant introducing some specific features of the environment.

1.3.1 Immersive audio reproduction

Since conventional systems are thought for obtaining the optimal acoustic sensation in a particular position of the listening environment (the so called *sweet spot*) and it is impossible to achieve a correct source localization with a limited number of loudspeakers, the performance of this kind of systems may be improved using immersive audio systems. To this aim, research efforts have focused on Wave Field Synthesis (WFS) that allows sound fields synthesis through loudspeakers arrays [39]. Similarly, Wave Field Analysis (WFA) implements a sound field recording technique based on microphones arrays [40]. Thus, these approaches allow to record the entire sound field in the recording room (WFA) and subsequently to reproduce it in the listening room (WFS) with an accuracy depending on the number of microphones and loudspeakers. As a matter of fact, novel advanced teleconferencing systems could be studied based on these immersive audio techniques. But the application of these concepts in this scenario needs the implementation of a real time solution to drive the loudspeakers during the audio streaming. Unfortunately, a straightforward implementation of these algorithms is not feasible due to the dramatically high number of inputs/outputs leading to unreasonable computational complexity. Therefore, studies of efficient solutions are essential for a real time implementation. Moreover, Wave Domain Adaptive Filtering (WDAF) has to be introduced for the application of adaptive algorithms to WFS and WFA [41]. The computational cost can be further reduced taking advantage of filter banks theory.

In this work, the aforementioned issues have been investigated and efficient solutions have been described for their real time implementations.

1.3.2 Listening environment equalization

Audio rendering algorithms are needed to improve the perceived audio quality. In particular, equalization represents a powerful tool capable of dealing with the frequency response irregularities. It can be applied in a teleconferencing system to make more natural the communication, compensating for speaker placement and listening room characteristics [42]. In the last decades, several efforts have been made; specifically regarding room equalizers, they can be divided in two categories: single position and multiple position equalizers. In single position room equalizers, the equalization filter is designed on the basis of a measurement of the impulse response (IR) in a single location [43]. Anyway, the impulse response varies significantly with the source and receiver positions in the room [44][45] and a single filter can equalize only the common trends of the room impulse response, producing poor improvements. One proposed solution to this problem is to select the most suitable equalization filter from a spatial equalization filter library whenever the receiver moves [46]. Furthermore, in the literature, a variety of approaches on multipoint room equalization are documented [42][47][48][49][50][51][52], enlarging the equalized zone through the room impulse response measurement in multiple locations.



Moreover, a still open problem in the field of room equalization is the derivation of effective and perceptually useful mixed-phase room equalizers for improving the objective and subjective quality of sound reproduction systems [42][53].

In this work, the analysis of a new multipoint magnitude equalization and a novel solution for combining a well known minimum-phase room equalization technique with a suitably designed room group delay equalizer have been presented.

1.4 Other contributions

The "missing-fundamental" phenomenon has been exploited for bass enhancement. In the wide field of audio devices, the use of large loudspeakers is not always possible according to size constraints (e.g., portable audio, laptop). This condition leads to the adoption of small loudspeakers characterized by a poor frequency response in particular at low frequencies. For this reason, many efforts have been made to enhance low frequency sound reproduction using algorithms based on the "missing-fundamental" principle [54][55]. In this thesis, a real time virtual bass algorithm based on high resolution spectral estimation

techniques has been discussed.

2 Research planning and activities

PhD activity has been organized starting from the state of the art of the chosen topics. Subsequently, the novel ideas have been investigated, first, from a theoretical point of view and then, through experimental tools, such as the Nu-Tech framework that allows to implement and test real time algorithms in multichannel scenarios [56].

Moreover, several international conferences were attended in order to present the novel contributions and to have a chance of relating with other researchers interested in the same topics (International Workshop on Acoustic Echo and Noise Control on Sep. 2008, 127th Audio Engineering Society Convention on Oct. 2009, and European Signal Processing Conference on Aug. 2010).

3 Analysis and discussion of main results

This work copes with several issues in the context of multichannel teleconferencing audio systems mainly concerning on multichannel acoustic echo cancellation. More specifically, the problem of channels decorrelation is taken into consideration, due to the ill-conditioning of the covariance matrix. Indeed, possible convergence problems may occur and a method to reduce the interchannel coherence has to be introduced. The novel idea consists in the reduction of the correlation between channels through a missing-fundamental-based approach that estimates and removes the fundamental frequency from one channel [1]. Audio quality and stereo perception are not affected because the human ear is capable of recreating the fundamental through the other harmonics. To this purpose, a notch filter, adapted at every new sample of the input signal, is exploited: its behaviour is controlled by the coefficient k_0 able to track the fundamental frequency and by the polezero contraction factor α controlling the filter's bandwidth:



$$H(z) = \frac{1 + 2k_0 z^{-1} + z^{-2}}{1 + k_0 (1 + \alpha) z^{-1} + \alpha z^{-2}}.$$
 (1)

A sigmoid function-based control for the adaptive algorithm has been introduced in order to speed up the fundamental frequency estimation, allowing even its more correct tracking [2], as described by the following equation:

$$k_0 = \frac{2}{1 + e^{-g_0}} - 1.$$
⁽²⁾

Figure 1 shows the fundamental frequency estimated by the adaptive notch filter for a speech signal sampled at 16 kHz. Several experimental results are carried out with the aim of validating the approach, according to interchannel coherence reduction, in terms of Magnitude-Squared Coherence (MSC), and convergence performance, in terms of misalignment, i.e., difference between the estimated and the real echo paths. The input signal is characterized by a very high initial interchannel coherence. Since the notch filter bandwidth varies with the different values of α , the relation between α and the interchannel coherence has been tested. Figure 2 shows how the coherence reduction is spread over the frequency range of interest and increases when the contraction factor is reduced. This result is confirmed by the behavior of misalignment. Figure 3 shows how an increased convergence speed is visible in the low-frequency range of interest.

Nevertheless, the missing-fundamental-based approach can be applied only to the lowfrequency band, where the fundamental frequency is typically contained. Thus, it is necessary to combine this technique with other techniques to guarantee stereo decorrelation over the whole spectrum. A first solution is proposed in [4] based on the combination of the missing-fundamental-based approach with another technique just existing in the literature [4]. Another solution is proposed in [5] according to the human sensitivity to frequency. Furthermore, listening tests were performed for proving audio quality and stereo perception preservation: the subjects assigned to all tracks under test a score in the excellent interval and all the directions have been correctly identified.

Since the application has to be enough efficient to be implemented on a real system, its real time performance has been tested on the NU-Tech framework. As shown in Figure 4, a collection of NUTSs (NU-Tech Satellites) has been realized in order to test it in a real time SAEC application [3].



Figure 1: Estimated fundamental frequency for a speech signal sampled at 16 kHz.





Figure 2: MSC for a speech signal $f_s=16$ kHz (I) without decorrelation and with (II) masked noise approach [30] and proposed approach (III) $\alpha=0.9$, (IV) $\alpha=0.6$, (V) $\alpha=0.3$, (VI) $\alpha=0.1$.



Figure 3: Misalignment for a speech signal $f_s=16$ kHz (I) without decorrelation and with (II) masked noise approach [30] and proposed approach (III) $\alpha=0.9$, (IV) $\alpha=0.6$, (V) $\alpha=0.3$, (VI) $\alpha=0.1$.



Figure 4: Real time implementation on a PC through Nu-Tech framework.



The aforementioned novel contributions for channel decorrelation represent the main issue among research activities. Other contributions have been obtained in the field of multichannel adaptive filtering algorithms, immersive audio reproduction, listening environment equalization and bass enhancement.

As regards multichannel adaptive filtering algorithms, a two-channel VSS-FDAF solution has been introduced for improving the convergence speed of adaptive filtering algorithms: experimental results have proved the effectiveness of the approach.

As regards immersive audio reproduction techniques and listening environment equalization, novel contributions have been obtained in these fields with the aim of improving the performance of conventional teleconferencing systems. In the context of WFA/WFS, real time implementations of WFS scenario on a PC [7][8][9][10], a WFS-based digital pointing of line arrays and its real time implementation [8][11], and a WOLA-based approach for WDAF with the aim of applying WFS technique to real scenarios [12] have been proposed, focusing on suitable processing for lowering the computational cost. Experimental results have proved the effectiveness of the approaches.

Further contributions have been obtained in the context of audio equalization, proposing the evaluation of a novel multipoint magnitude equalization system [13][14], that considers different prototypes and inversion algorithms, and introducing a group delay compensation [15], for improving audio quality. Test sessions have proved the effectiveness of the approaches.

Finally, the missing-fundamental approach has been exploited for a new bass enhancement solution [16].

4 Conclusions

Research activities focused on multichannel teleconferencing audio systems. In particular, the study of the missing-fundamental phenomenon has brought to novel contributions for channel decorrelation that plays a significant role in the context of multichannel acoustic echo cancellation. Tests results show that these proposed approaches outperform the techniques existing in the literature. Moreover, a two-channel variable step-size adaptive filtering algorithm has been proposed. Since the performance of teleconferencing systems can be greatly improved through immersive audio techniques and audio equalization, novel contributions in these fields have been discussed. Eventually, the missing-fundamental phenomenon has been exploited for the development of a new approach for bass enhancement. Future works could be oriented towards the application of these new solutions to real scenarios.

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