

Topics in Acoustic Echo and Noise Control

Selected Methods for the Cancellation of Acoustical Echoes, the Reduction of Background Noise, and Speech Processing

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Acoustic Echo and Noise Control – Where did we come from and where are we going?

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The invention of the telephone about 150 years ago extended the range of verbal communication between humans beyond the bounds given by the power of their voices. Using this technology, however, was – and still is – inherently connected with some inconveniences. The talkers have to hold a handset such that the loudspeaker is close to their ear and the microphone is adjacent to their mouth. Even then, speech quality is reduced and ambient noise may be picked up. Replacing the handset by a microphone and a loudspeaker now positioned a – short – distance from the talker increases the loss of the transmission loop by say 20 dB [6]. Furthermore, the level of ambient noise collected by the microphone is increased and the echo from the loudspeaker signal is picked up.

Methods of acoustic echo and noise control aimed to remedy these disadvantages exhibit a long history. Originally, efforts were focussed on the development of hands-free telephone systems. In the following section we will highlight some of the important steps toward today's systems. We will also point out that the developments were always linked to the technology available at the time of their proposal.

This introductory chapter will close with an overview of important current developments reported in the following chapters of this book in detail by international experts in the field of acoustic echo and noise control.

1.1 The Journey to Maturity

1.1.1 The Problems to be Solved

To restore the comfort of a face-to-face conversation over a hands-free telephone connection three major problems have to be solved:

- Comfortable volumes of the speech signals have to be provided for both partners without destabilizing the electro-acoustic loop.

- The echoes of the loudspeaker signal(s) picked up by the microphone(s) have to be reduced to an acceptable level without affecting double talk performance.³
- Ambient noise has to be removed from the microphone output signal(s) to below a level that might be tolerable in case of binaural listening.

If there is only one hands-free telephone (locally) and the remote talker uses a handset it is this party who suffers mostly from inappropriate solutions to the problems mentioned above. The local talker may move the loudspeaker closer to his ear or increase the volume – increasing the risk that the loop starts howling. His echo and noise problem is minor at most since his partner holds the microphone close to his mouth. Consequently, when algorithms for echo and noise control are designed, most attention has to be paid to the situation of the remote communication partner.

Historically, when efforts to stabilize the electro-acoustic loop started, only classical acoustic means were available. Loudspeakers and microphones in separate units or combined in one housing were put in favorable positions. Furthermore, the walls, floor, and ceiling of the enclosing room had to be treated with absorptive materials [4].

It was not until the 1950s that signal processing means could be considered. Voice controlled switching of the receiving and sending circuit, center clipping and frequency shifting were employed at that time.

1.1.1.1 Voice Controlled Switch

Voice controlled switching (see Fig. 1.1) means that either the receiving or the sending line is interrupted [3]. Thus, only half-duplex communication is provided, double talk is impossible. Proper control of the switching is difficult. It is based on the *estimated* activities of the incoming and the outgoing line. Noise and echoes can cause malfunctions. The beginning and the end of utterances may be “chopped off”. For the “inactive” partner of a conversation it is not possible to break in.

A considerable number of modifications to the loss control circuit have been proposed over the years. Instead of switching lines completely on and off, a finite attenuation is inserted and is distributed on the incoming and the outgoing circuit according to the estimated activities. Short-term power estimations – utilized for speech activity detection – with different time constants can improve the performance at the starts and at the ends of words.

All these modifications can reduce, but not completely remove, the problems described above. Nevertheless, voice controlled switching is still used in modern echo control systems. In an environment with adaptive circuits, minimum levels of echo attenuation as required by international standards can only be guaranteed by such circuits. However, only the difference of the

³ The term *double talk* describes periods in which both – the local and the remote – communication partners speak at the same time.

attenuation already provided by echo cancellation and/or echo suppression (see below) and the one called for by the standards has to be inserted. Thus, in these cases, the impact of loss control on speech quality may be hardly noticeable, if at all.

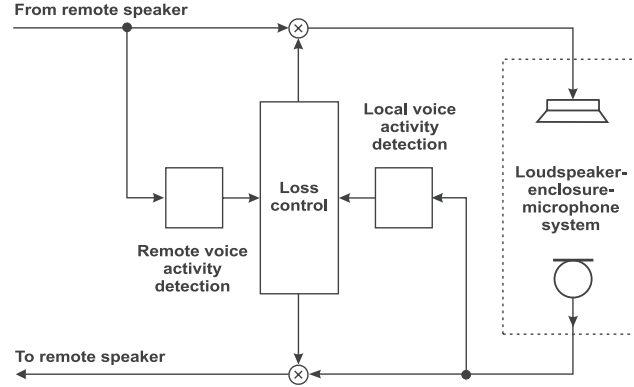


Fig. 1.1. Principle of a loss control circuit.

1.1.1.2 Center Clipper

A center clipper (see Fig. 1.2) inserted into the transmission circuit suppresses small output signals [2]. If these signals contain only the acoustical echo – plus some small ambient noise – the echo is removed completely. However, if the echo is superimposed onto a local speech signal, the center clipper proves to be ineffective and only distorts the speech signal. Again, a large number of modifications – including adaptive thresholds and adaptive slopes – have been proposed over the years. Nevertheless, the use of center clippers in acoustic echo and noise control seems to remain a makeshift solution.

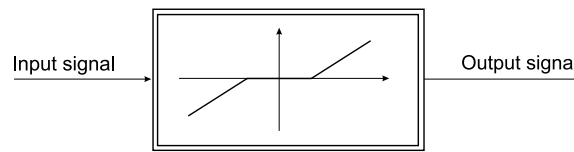


Fig. 1.2. Center clipper.

1.1.1.3 Frequency Shift

The magnitude of the transfer function of a typical loudspeaker-enclosure-microphone (LEM) system exhibits a sequence of maxima and minima with a separation of 5 to 10 Hz (see Fig. 1.3). Peaks and valleys are, respectively, about 10 dB above and below the average magnitude. Based on this observation a *frequency shift* of the loop signal can increase the stability margin [16]. This method was proposed especially for systems like public-address systems where the loudspeaker output signal feeds back directly into the talker’s microphone. It can be used in hands-free telephone applications as well. Its primary component is a single-sideband modulator that performs a shift of the loop signal by a few Hertz. Thus, stationary howling can not build up. It is moved to higher or lower frequencies – depending on whether the modulation frequency is positive or negative – until it “falls” into a minimum of the transfer function of the LEM system.

In speech communication systems frequency shifts of about 3 to 5 Hz are scarcely noticeable. The stability gain achievable with this method depends on the signal and the acoustical properties of the enclosure. For speech signals and rooms with short reverberation times, the stability gain is of the order of 3 to 5 dB; for rooms with long reverberation times it can go up to about 10 dB [15].

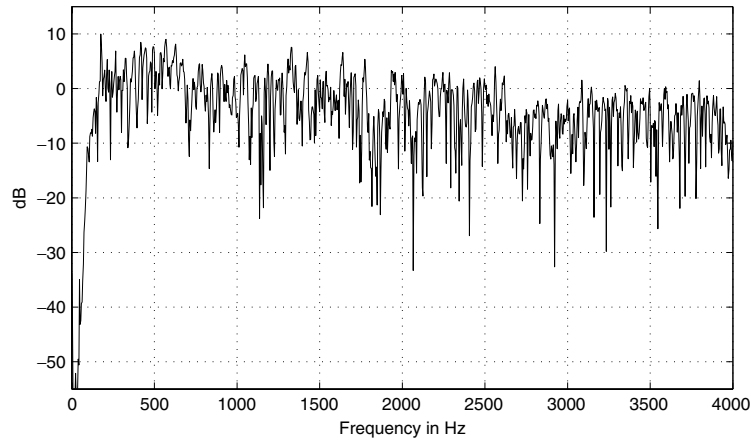


Fig. 1.3. Absolute value of a transfer function measured in a small lecture room.

1.1.1.4 Echo Cancellation and Echo Suppression

The invention of the least mean square (LMS) algorithm in 1960 [19] can be considered as the most important development for adaptive filtering. This

procedure became the “work horse” for today’s existing enormous variety of algorithms for filter adaptation. Its numerical complexity is proportional to $2N$, where N is the number of filter coefficients. Given a proper step size, it does not cause stability problems. However, its speed of convergence is low especially in case of correlated inputs like speech signals.

The potential of the LMS algorithm for echo cancellation or suppression was recognized soon after its publication. The first application was the cancellation of *electrical* echoes on long distance transmission lines [11, 17]. Compared to *acoustic* echoes, line echoes are considerably shorter. Thus, they require less complex filters. In contrast, the processing of acoustical echoes necessitates adaptive filters that are extremely demanding with respect to signal processing power. It is, therefore, not astonishing that the application of adaptive filters to acoustic echo and noise control was not considered before the late 1970’s [14]. Even at that time the signal processing technology to implement those filters could only be seen on the distant horizon.

Simulations and laboratory experiments in the 1980’s affirmed the weakness of the LMS algorithm with respect to correlated – e.g. speech – signals. These results started a strong effort by researchers to utilize the recursive least squares (RLS) algorithm for acoustic echo processing. In contrast to the LMS algorithm, the complexity of this procedure grows quadratically with the number N of filter coefficients that have to be adapted. It can handle correlated signals very well since it has a “built in” decorrelation facility. This, however, needs the inversion of the short-term $N \times N$ correlation matrix of the input signal. In the applications considered here, N may range up to the order of several thousands. This matrix can become singular by the nature of the input signal or the estimation procedure. As a consequence, the RLS algorithm frequently becomes instable for echo processing. The stabilization and reduction of the complexity to a linear dependency on the number of filter coefficients was one of the major topics at the first *International Workshop on Acoustic Echo Control* held in 1989 in Berlin, Germany. Despite all the efforts at that time and in the following years, the problems of applying the RLS algorithm to acoustic echo cancellation still seem unsolved. The situation can be highlighted by a cartoon (see Fig. 1.4).

The LMS and the RLS algorithms may be considered extremes in the world of adaptive algorithms. This holds with respect to complexity and numerical problems, but also with respect to their dependence on past signals and settings of the filter coefficients. The LMS algorithm uses only current inputs whereas the RLS procedure looks back on past inputs according to a forgetting factor. In order to stabilize the RLS algorithm it may be necessary to furnish the algorithm with a long memory. This turns out to be a handicap when changes to the LEM system have to be tracked.

The affine projection (AP) algorithm [13], and especially its fast version [9], provides a good compromise between the LMS and RLS algorithms. Compared to the LMS algorithm, numerical complexity is modestly increased. The speed of convergence for speech inputs nearly reaches that of the RLS procedure.



Fig. 1.4. “Better to have a sparrow in the hand than a pigeon on the roof” (by Prof. Helmut Lortz).

These properties are achieved by optimizing the filter coefficients not just with respect to the current input signals – as the LMS algorithm does – but also optimizing for $M - 1$ preceding inputs. M is called the order of the algorithm. For $M = 1$ it is equal to the LMS procedure. Like the RLS algorithm, the AP method needs the inversion of a matrix. This, however, is of size $M \times M$ only. For speech inputs M can be chosen in the order of 2 to 5. By comparison, in this situation the RLS algorithm would require inversion of an $N \times N$ matrix with N in the order of 1000.

1.1.1.5 Echo Cancellation

Echo cancellation is achieved by using the output of a filter that attempts to match the LEM system (see Fig. 1.5). Since the latter is changing constantly, the filter has to be adaptive.

During the development of echo cancellation filters (ECFs), a long-winded discussion took place whether a transversal (FIR) or a recursive (IIR) filter is better suited to model the LEM system. Since a long impulse response has to be modelled by the ECF (see Fig. 1.6), an IIR filter seems best suited at first glance. However, upon further inspection, the impulse response exhibits a highly detailed and irregular shape. To achieve a sufficiently good match, the replica must offer a large number of adjustable parameters. Several studies have shown that an IIR filter does not provide a sufficiently large advantage over an FIR filter to justify the enormous cost of controlling its stability [10, 12, 20]. The even more important argument in favor of an FIR filter is that adaptation algorithms for FIR filter are available and that the stability of these filters need no extra control.

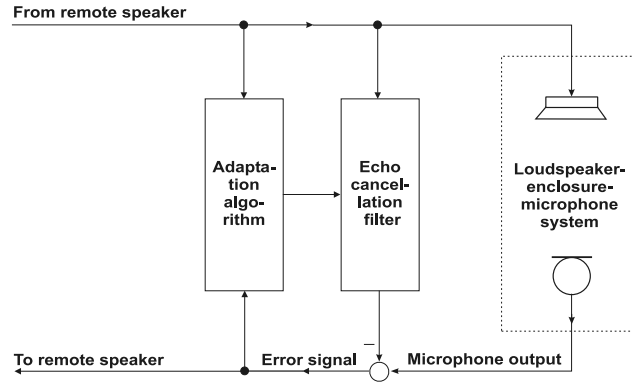


Fig. 1.5. Principle of echo cancellation.

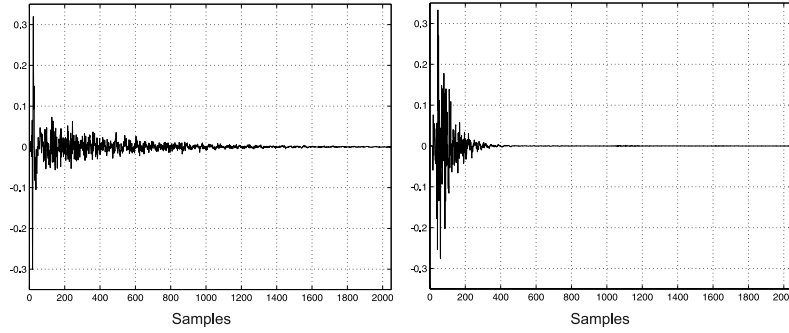


Fig. 1.6. Impulse response of LEMs measured in an office (left) and in a car (right). The sampling rate is 8 kHz.

1.1.1.6 Control of the Filter Adaptation

From a control engineering point of view, the adaptation of the echo cancellation filter is equivalent to the identification of a highly complex system. To make things even more difficult, the adaptation has to be performed in an environment where the signal-to-noise ratio often falls below 0 dB. A short example may help illustrate the complexity of the task: Assume, that the error signal (see Fig. 1.5) suddenly rises. This can have two reasons:

- The local speaker started talking or a local noise started.
- The local speaker changed his position and thus changed the impulse response of the LEM system.

The control of the adaptive filter that can only rely on the output signals of the microphone and the ECF cannot distinguish between the two cases. The

reactions, however, have to be diametrical: Adaptation has to be frozen in the first case whereas it has to be opened as much as possible in the second instance. No algorithm for adaptive filters can handle this situation without additional information.

This information has to come from estimates of various quantities. Most of them are not directly measurable. Independent of the currently available processing power and processing speed, the reliability of these estimates depends critically on the length of the signal segment the estimation can be based on. This simply means that it may be necessary to delay control actions until dependable estimates are available. In cases where erroneous control signals lead to a rapid divergence of the filter coefficients – as in the first case of the example given above – rapid actions based on temporary estimates are necessary to prevent “dangerous” situations.

In this respect, the question of applying adaptation algorithms that result in a high speed of convergence of the filter coefficients becomes an additional consideration: Fast adaptation requires a reliable and fast acting control structure. The reaction time of the latter, however, is limited by the time necessary to acquire a sufficiently long signal segment. If this condition is not fulfilled, an algorithm not reacting “nervous like a race horse” may lead to better results.

1.1.1.7 Echo and Noise Suppression

With echo cancellation, the achievable echo attenuation is limited to at most 30 dB in an ordinary office. This is due to thermal fluctuations [5], nonlinearities within the A/D and D/A converters, within the electro-acoustic converters [18], and, lastly, the insufficient length of the echo cancellation filter. To improve echo attenuation, a filter in the transmission circuit is necessary (see Fig. 1.7). The transfer function of this filter is adapted according to the spectrum of the speech signal. A similar filter can be used to suppress ambient noise picked up by the microphone. The coefficients of both filters, however, have to be adapted according to the different properties of the residual echo and the noise.

In contrast to echo cancellation by a filter parallel to the LEM system, echo and noise suppression affects the quality of the transmitted speech signal. Therefore a compromise between speech quality and echo and noise suppression is always necessary.

1.2 State of the Art

Acoustic echo and noise control are among to the most challenging problems in digital signal processing. Many authors confirm this statement. The effort in research and development over the last three decades has been overwhelming. As a result, the problems around “classical” single channel hands-free systems are very well understood and are basically solved. There are systems available

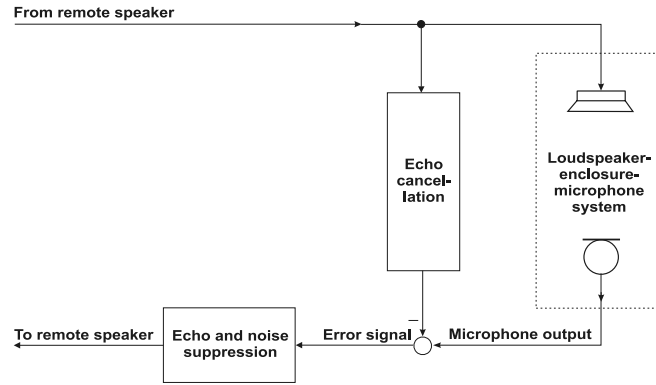


Fig. 1.7. Principle of echo and/or noise suppression.

that function satisfactorily. The fact that systems with poor performance are still in use seems to depend on two reasons: The sales price of consumer products is calculated by the cent. The hands-free functionality has to be implemented with absolutely minimum cost. Furthermore, the benefit of a high-quality echo and noise control system is with the remote communication partner; and he is not the one who pays for the system.

Stereophonic systems still offer open questions. Due to the fact that both signals may be fully dependent on each, the optimal settings of the coefficients of the ECFs are not unique. A remedy is found by artificially distorting one or both signals [1]. Fortunately, real systems behave well and only small distortions are necessary. From a conceptual point of view, a more “elegant” solution seems desirable. Promising approaches towards this goal are the subject of current research.

Parallel to the growth of processing power, new applications that require more and more sophisticated systems move into the field of vision of researchers. Advanced methods are based not only on measured and estimated signals but also on expert knowledge of the underlying processes.

New algorithms for filter adaptation are proposed that are tailored to the specific properties of the echo and noise control process.

Multi microphone and/or multi loudspeaker configurations offer additional degrees of freedom to the echo and noise control problem. Microphone arrays allow the speaker to be located and tracked. The same holds for noise sources. All these methods aim to improve the signal-to-noise ratio of the audio signal. With loudspeaker arrays, radiation patterns are generated such that microphones are located within the minima of sound intensity. Both array approaches are able to reduce the echo problem. Since the electro-acoustic properties of inexpensive microphones exhibit considerable variances, automatic scaling improves the performance of microphone arrays.

Methods of sound source separation isolate individual speakers from mixtures of speech signals and noise. Blind methods for this task are under investigation.

The application of the Kalman filter offers considerably better results in noise reduction than “classical” procedures like spectral subtraction. Processing in subbands overcomes the complexity problem of the filter.

Based on models for speech production and on the properties of human sound perception, methods for enhancing speech signals beyond the quality of the still widely used telephone speech are proposed. Making use of code books for narrow and wide band speech signals and of the masking properties, it is possible to fool the human ear such that the impression of listening to wide band speech is generated from processing narrow band speech signals without using any side information.

The deverboration of speech signals and the inclusion of dictionaries improves the reliability of speech recognition systems.

The development of new methods is – at least partly – related to new application areas. The technology for implementing demanding algorithms in hearing aids has become available only in the last few years. Digital implementations allow for more adjustable parameters. Thus, a better match to hearing impairments is possible.

Demands for greater passenger comfort initiated the search for solutions for in-car passenger communication.

This list is far from complete. With a look into the proceedings of recent signal processing conferences and into signal processing journals it can easily be extended.

The quality of early real-time implementations of echo and noise control systems was bounded by the capacity of signal processors available at that time. This limitation no longer exists. Forgetting for a moment the above remarks on the allowed costs of consumer products, algorithms implemented now or in the near future can perform at their theoretical limits.

The availability of powerful general purpose computers and high level simulation tools allow the simulation of algorithms for acoustic echo and noise control with low effort. Reality, however, turns out to be much more complex than even sophisticated models. Results based only on simulations should be handled with extreme care.

1.3 Outline of this Book

It is the purpose of this book to describe a number of highly important developments in acoustic echo and noise control in more detail. Distinguished authors present overviews and results of their research. Their contributions are organized in four Parts focusing on multi-microphone processing (Part II), echo cancellation (Part III), noise reduction (Part IV), and selected applications (Part V).

In Part II, Chapter 2 addresses the problem of time-varying echo paths, high level background noise, and frequent double-talk. A new joint acoustic echo canceller and beamformer is derived and evaluated. The advantages of the joint system are shown by a realization that integrates a stereophonic echo canceller and a generalized sidelobe canceller. The described solution requires only one echo canceller for an arbitrary number of microphones and no separate adaptation control.

Chapter 3 treats the problem of separating multiple sources of audio signals – as occurs during teleconferences or in hearing aids, to name only two application areas. The input signals to several microphones are convolutive mixtures of speech signals and ambient noise. The solution uses blind independent component analysis in the frequency domain. The phenomena of permutation and circularity are addressed and successfully solved. The authors present a complete solution for source separation. Experimental results are included.

In Chapter 4 techniques for the localization of acoustic sources are presented. The methods are based on only two microphones and perform a precise time-delay estimation. Using these techniques, a moving speaker can be tracked and the direction of high sensitivity of a microphone array can be steered such that it points towards this speaker.

Part III starts with adaptation algorithms for filters that have to model systems with sparse impulse responses (Chapter 5). Sparse in this context means that only a small percentage of the sample values of the impulse response of the original system exhibits values significantly larger than zero. Based on this prior knowledge, general procedures for filter adaptation like the LMS or the RLS algorithm can be considerably improved. It is shown how refined algorithms can be derived and how known procedures and the algorithms developed in this chapter are related to each other. Further, the distinction between algorithms with linear and with nonlinear updates is made. In the case of acoustic echo cancellation, procedures with nonlinear updates can be advantageous.

Cancellation of acoustical echoes needs the update of filters with up to several thousands of coefficients. In Chapter 6 it is shown that the computational complexity associated with this task is reduced by updating only a fraction of the coefficients at a time. Through proper selection the performance of the filter degrades only by a small degree. The sorting of the filter coefficients, however, may lead to a considerable overhead. Fast sorting algorithms described in this chapter overcome this problem. It is also shown that selective update methods may be a remedy against the misalignment of the ECFs due to their non-uniqueness in stereophonic systems.

Realistic electro-acoustic echo paths may contain a number of elements with non-negligible nonlinearities, such as low cost loudspeakers, overloaded amplifiers and non-ideal converters. In Chapter 7 a nonlinear model of the echo path is formulated consisting of a cascade of linear and nonlinear filters. It is explained that second order Volterra filters are suited to model loud-

speaker nonlinearities, whereas power filters are proper models of memoryless nonlinearities as they occur, for instance, with overloaded amplifiers or low cost converters. Adaptation algorithms for both filter types are developed and applications to real systems are discussed.

Systems for acoustic echo and noise control require sophisticated procedures to supervise all subsystems in order to avoid performance degradations in case of “dangerous” communication events. Incidents like these are, for example, the sudden onset of double talk or changes in the echo path by movements of the local speaker. To arrive at a robust control structure, the outputs of detectors and estimators have to be combined in an intelligent way. Chapter 8 presents several systematic approaches for this combination. Their additional computational effort is made up for by an improved overall system performance.

Chapter 9 in Part IV gives an in-depth treatment of noise reduction algorithms. The emphasis is put on single microphone solutions based on Wiener filtering and spectral subtraction. The design of Wiener filters is described in the time and in the frequency domain; filtering effects and realisations are explained. A second focal point is spectral subtraction methods and their relationship to Wiener filters. Central to both methods is the estimation of the noise power spectral density that is discussed in the following sections. Finally, techniques for the design of uniform and non-uniform filter banks – including wavelets – are described.

In Chapter 10 a Kalman filter based single channel noise reduction method is presented. It starts with an analysis of speech signals and car noise and the formulation of parametrical models needed for the Kalman filter. A special procedure for the estimation of speech parameters from noisy signals is developed. The complexity problem of the Kalman filter is overcome by a subband approach. Methods are discussed to enhance the noise reduction performance of the filter. A comparison with more conventional noise reduction methods closes this chapter.

Part V opens with considerations about the assessment of the quality of acoustic echo and noise control systems. Subjective listening tests are the most reliable means. To perform such tests, however, is time consuming and expensive. Therefore, especially during the algorithm development phase, the availability of objective tests is desirable. In Chapter 11 both classes of tests are discussed. It is described how they can be performed and how their results have to be evaluated.

Chapter 12 is concerned with auditory scene analysis. This technique is inspired by the ability of humans to segregate a sound source from a mixture of multiple sources even from only a monophonic signal. A system for computational acoustic scene analysis (CASA) performs successively four subfunctions. Firstly, a peripheral analysis is performed where the auditory scene is decomposed into a time-frequency representation. A feature extraction follows. These features provide the basis of a segmentation and, finally, of a grouping. Here, segments for the sound source of interest – the target – and

the interferers are created. Finally, the waveform of the target is synthesized from the related segments. The approach described here is primarily feature-based. Except for unvoiced grouping, no prior knowledge is assumed.

Chapter 13 deals with the synthesis of wave fields, a novel method for spatial sound reproduction. It applies arrays of large numbers of loudspeakers to recreate a sound field in a listening area. Even if the main applications of this techniques are in the areas of entertainment and performing arts, it may also be used to recreate sound fields for human communications. The technique is based on the physical properties of wave propagation. It applies the solution of the acoustic wave equation by Green's function. Signal processing methods to derive the input signals for the loudspeakers of the wave field synthesis system are reported. Exemplary implementations close this contribution.

Chapter 14 deals with so called in-car communication systems. They help to ease communication between passengers in a car. Such a system is especially helpful for passengers seated in the back of the car to understand those seated in the front. The problem that has to be solved is comparable to the one present with public address systems where the electro-acoustic loop is closed within the enclosure and where only very short processing delays are tolerable. In contrast to e.g. hands-free systems, the (local) speech signal and the echo signal are highly correlated. Therefore, new control structures have to be developed. Since standardized quality measures for in-car communication systems do not yet exist, measurements and subjective test are also reported.

The continuous improvements in semi-conductor technology allowed the changeover from analog to digital technology. Chapter 15 describes algorithms implemented in high-end hearing aids to improve the hearing ability and the hearing comfort of impaired people. The procedures have to take into account the special requirements of these devices. For example, the loudspeaker and the microphones are very close together and a high amplification is inevitable. Furthermore, different listening situations call for their automatic classification, enabling the selection of different parameter sets.

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