A course on active noise control held at the Technical University of Berlin

André Jakob

Technische Universität Berlin, Institut für Technische Akustik, Einsteinufer 25, D-10587 Berlin, Germany

Email: andre.jakob@tu-berlin.de

Introduction

During the summer semester of 2002 a course on active noise and vibration control was held at the Institute of Technical Acoustics of the Technical University of Berlin and was repeated in the winter semester of 2002/2003. The course was designed for students of acoustics. It introduced the physical basics of active control as well as control strategies, i.e. feedforward, feedback and adaptive digital filters, which are commonly used for control.

Contents of the course

The course is divided into three main parts and is organized as follows.

1. Introduction, state of the art, example applications,
2. Active noise and vibration control from a theoretical point of view,
3. Digital signal processing, adaptive digital filters.

Part 1

In part 1 the participants are taught what active control is about and some applications investigated in the past are briefly discussed. The principle of destructive interference is easily understood by the students, e.g. by showing the well-known figures from the patents of Paul Lueg in 1936. But, the limits of active control are not usually obvious for beginners.

Some applications are presented, e.g. active noise control (ANC) in cars, in aircrafts (ASANCA-project) and ANC as exhaust noise damping systems (”active industrial silencer”), active vibration control (AVC) in truss structures, active isolation of rotating machines and, the well-known commercially available headphone with ANC.

The size of current active control research is easily demonstrated by the reference bibliography made of Dieter Guicking [1] which contains more than 8300 references.

Part 2

The large Part 2 contains a lot of analytical calculations. These are:

Necessary accuracy of the secondary signal Superposition is calculated of two sine-functions and the residual amplitude is displayed depending on the amplitude and phase error.

Plane waves ('active reflector/active absorber') A plane wave in a channel can be cancelled out downstream by the sound field of one secondary source. This results in the active reflector with a standing wave upstream. A second source is used to prevent this standing wave upstream which results in the active absorber. Both are illustrated with MATLAB-movies.

Hermitian Quadratic Form This special form of quadratic vector equation is used in all the following examples. The minimum is derived.

Monopoles in free field The sound field of two or more monopole sources is calculated in the far field and optimal secondary sources for minimizing the total radiated sound power are determined and discussed in terms of the distance between the secondary sources and the primary source. Four special cases are then illustrated with animated MATLAB-movies, i.e. two equal sources drifting apart, two equal sources but with opposite sign drifting apart, a primary source and an optimally adjusted secondary source drifting apart and a primary source and two optimally adjusted secondary sources drifting apart.

Active control of enclosed sound fields Beginning with the partial differential equation for sound waves in air an actively controlled three-dimensional closed room is calculated. The primary and secondary sources are point sources somewhere in the room. The objective of minimization is either the mean squared sound pressure level inside the room, i.e. when global control is assumed, or, the sum of squared sound pressure levels over some positions in the room, i.e. when local control is assumed. The results are given in terms of frequency responses with and without control and are illustrated at some selected frequencies as shown in Figures 1 to 3. It is deduced that it is often better to use more error microphones than loudspeakers. It is also shown what happens if one places the error microphones such that only some regions of the room are to be controlled, i.e. large level reductions in the controlled region but also eventually large deteriorations in the uncontrolled regions.

Active vibration control of a plate strip Beginning with the partial differential equation for bending waves in plates, an actively controlled one-dimensional plate strip is calculated. The primary and secondary sources are vibrational inputs described by a force perpendicular to the plate strip. The results are discussed in terms of number and position of secondary sources and accelerometers. Global and local minimizations are compared.

Active structural acoustic control of a plate strip As an extension to the former item the radiated sound power is minimized by adjusting vibrational inputs. The result leads to the well known concept of 'modal restructuring' and is discussed and compared with the minimization of the plate strips vibration itself.

Part 3

The topics of Part 3 are signal processing and adaptive filtering:

Experiment with manual control In one lecture an experiment is prepared in which the students adjust the secondary sources manually. A primary loudspeaker and a secondary loudspeaker are installed together with an error microphone and the students adjust an amplifier and a phase shifter in order to decrease the error signal to the minimum. This is quite an easy task. When an additional secondary loudspeaker and an additional error microphone is installed the students have to manually adjust two amplifiers and two phase shifters which is a much more tedious task. From these experiments the students get a feeling for the complexity of active noise control applications. In another little experiment the students use a second function generator and manually try to generate exactly the same frequency as the primary signals frequency. At this point the students see that this is nearly an impossible task.

Signal processing overview Here a very brief introduction to signals and systems is given. Rather than providing all the detailed mathematics an overview is presented comparing e.g. continuous-time signals vs. discrete-time signals, Fourier transform vs. DFT, periodicity in time vs. periodicity in frequency, Laplace transform vs. z transform, and so on. Aliasing and the need for anti-alias filter and reconstruction filters is explained and demonstrated by a little experiment: a DSP-board with bridged reconstruction filters is used to produce a sine wave with non-harmonic components according to the alias-frequencies around multiples of the sampling frequency. The students can see the stairway-like non filtered signal, they can hear the non-harmonic frequencies, especially when sweeping the frequency, and, they can observe the alias-frequencies around multiples of the sampling frequency on a FFT-analyzer.

Digital filtering FIR and IIR filters are introduced and the effects of real and/or complex zeros and poles are explained. It is shown that for the FIR filter two coefficients are necessary to exactly adjust each frequency in amplitude and phase.

Adaptive filtering After an introduction to optimal filtering leading to the Wiener filter, the LMS algorithm is derived for system identification. The extension to the filtered-x LMS algorithm is shown mainly graphically. Last, the multiple error LMS algorithm is developed.

Feedback adaptive filtering The internal model control (IMC) structure is derived from the block diagram. The feedback algorithm based on the feedforward filtered-x LMS algorithm extended with the reference signal estimator is given for the single channel case and for the multi channel case.

DSP An introduction to the structure of Digital Signal Processors (DSP) is given and, an example program for a FIR filter for a special DSP chip is explained.

In the last lecture a demonstration is given of the actively controlled double-glazed window investigated at the institute. The effects of control can be experienced and the adaptive filters can be observed working.

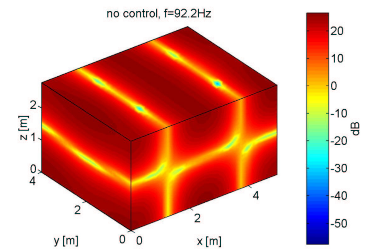

Figure 1: Sound pressure level inside room for a distinct frequency, no control, arbitrary units.

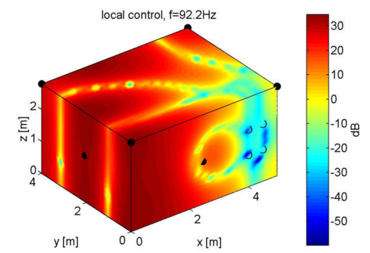

Figure 2: Local control, 8 loudspeakers (bullets), 8 microphones (circles), arbitrary units.

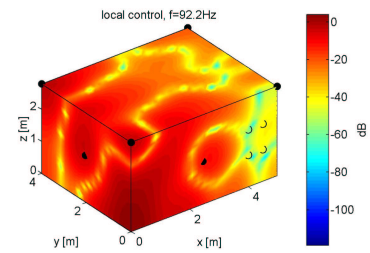

Figure 3: Local control, 6 loudspeakers (bullets), 8 microphones (circles), arbitrary units.

References

[1] Dieter Guicking, GORBI - Guicking's Online Reference Bibliography, Version 1.1, January 2003