SOUND-SYSTEM DESIGN FOR A PROFESSIONAL FULL-FLIGHT SIMULATOR

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ABSTRACT

In this paper, we present a sound system to be integrated in an accredited realistic full-flight simulator, used for the training of airline pilots. We discuss the design and implementation of a corresponding real-time signal-processing software providing three-dimensional audio reproduction of the acoustic events on a flight deck. Here, the emphasis is on an aircraft of a specific type. We address issues of suitable data acquisition methods, and, most importantly, of functional signal analysis and synthesis techniques.

1. INTRODUCTION

In a multidisciplinary research project, we are designing a flight simulator the behavior of which should be as realistic as possible. In this context, we aim at meeting the strict requirements of the highest international standard for full-flight simulators [1].

As can be seen in Figure 1, the flight simulator includes a realistic copy of the flight deck mounted on a motion base. Within certain limits, this allows for movement in all three dimensions. The visual illusion is provided by a projection system with an appropriate vision dome. Last but not least, the sound system is responsible for a lifelike reproduction of the sound field.

1.1. Embedding of Sound System in Flight Simulator

Topologically, the flight-simulator system consists of a network of independent modules managed by a central control unit (Figure 1). In general, these modules compute output parameters from a set of relevant input parameters (computation of the flight path; computation of the behavior of the engines). Conversely, the task of the sound-system module is the reproduction of a sound-field in reaction to input parameters. These input parameters may be continuous (true air speed, height), or binary (ground contact on landing gear).

More details on the concept and the structure of the sound system can be found in [2].

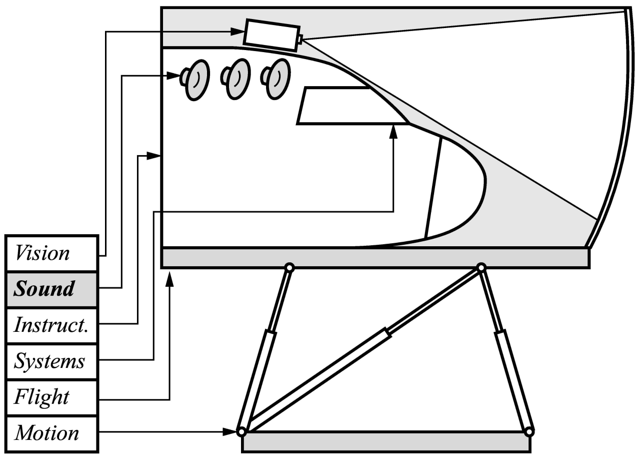

Figure 1: Scheme of Flight Simulator, including Sound System

1.2. Audio Hardware

In our sound reproduction system we employ half a dozen high-quality full-range wall loudspeakers mounted inside the cockpit. We have chosen their number and positioning in conformity with constructional as well as with acoustical requirements. The loudspeakers are distributed as uniformly as possible across the walls of the flight deck of the simulator.

In addition, the sound system includes a powerful subwoofer in order to intensify the illusion created by the simulator.

When choosing the audio hardware for the sound system, we had to consider that the simulator's motion base can produce high accelerations (up to $4\,\mathrm{g}$ in all three directions), which the loudspeakers will be exposed to.

1.3. Software and Computer Hardware

The signal processing part of the sound module is implemented using the platform-independent open-source software *Pure Data*, a graphical programming environment for real-time audio signal processing developed by Miller Puckette [3]. The sound simulation runs on today's standard PCs.

1.4. Objective and Subjective Tests

By means of objective measurements we will demonstrate that the sound produced by our system complies with the situation in a real cockpit in regard to (1) true sound pressure level, (2) frequency range, (3) localization of discrete sources, and (4) degree of spatial correlation.

Furthermore, we are continually validating our results with experienced pilots in order to match the overall sound experience presented by the simulator as closely as possible to the situation in a real cockpit.

2. ACOUSTIC EVENTS

In this section, we describe the most significant sounds that typically occur in the flight deck of an aircraft. On the basis of quite simplistic models, we deduce potential dependencies on physical parameters. In the target simulator, such parameters will constitute the input into the modules that are responsible for the creation of the different sounds. However, the sound of the aircraft of the specific type we are simulating need not depend on all parameters presented here. Actually, some sounds cannot even be heard at all.

We know from experience that most of the parameters change rather slowly (on the order of 100 ms). As a consequence, the sounds can be considered (quasi)stationary.

2.1. Continuous Sounds

Some acoustic events on the flight deck depend on parameters that are updated continuously in the simulation, i. e. at fixed time intervals. Such acoustic events are called "continuous sounds". This does not imply, however, that continuous sounds are always audible during normal flight conditions. The continuous sounds are:

2.1.1. *Noise Due to Air-Current*

In principle, the shape of the airframe of the aircraft does resemble streamlines. However, the shape is not perfectly smooth, but divided into sub-sections for technical reasons, e. g. the glasses of the front windshield. So, whenever there is relative movement between an aircraft and the surrounding air, turbulences emerge at these discontinuities. In acoustical terms, the turbulences result in a spectrally shaped noisy sound that seems to originate from where the front windshield and the body of the aircraft join.

Theoretically, the noise due to air-current should display dependencies on speed and on the direction of arrival of the airstream relative to the aircraft. This is in turn influenced by the aircraft's attitude as well as by the direction of potentially present winds. Moreover, the sound depends on air density, which varies with altitude and air temperature.

Figure 2 shows the sound pressure level for two different flight states. In both states, the energy seems to be concentrated in low-frequency regions. Note that the plots corresponding to the different states are similar yet clearly distinct.

On flight decks of the aircraft type we are simulating, the air-current noise is the predominant acoustic event.

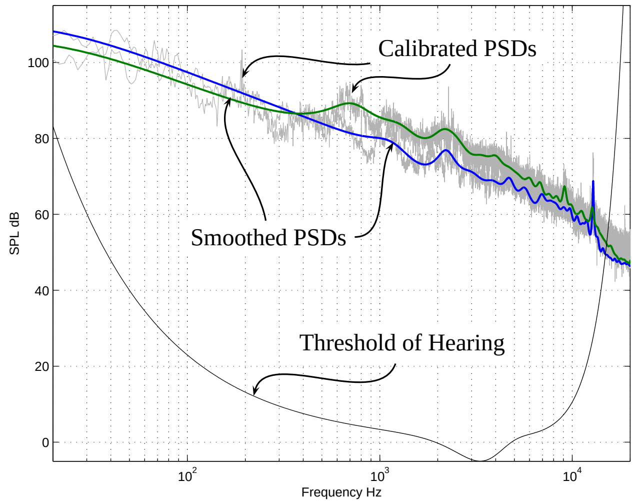

Figure 2: Calibrated Power Spectral Density (PSD) estimates (Welch's method of averaging modified periodograms) of the true sound pressure level for two different flight states. The smoothed spectra are computed by means of linear prediction analysis. For reference, the threshold of hearing is also plotted. The distinct peaks in the PSD estimates are discussed in Sections 2.1.2 and 2.1.6.

2.1.2. *Engine Noise*

Some noise evolves from the moving parts (i. e. rotor blades) present inside the engines of an aircraft. A specific rotor produces a harmonic sound with a fundamental frequency determined by its rotary speed. Note that in contrast to the noisy sound due to the air-current, the sound from the engines has distinct sinusoidal components.

Furthermore, the engine sound is affected by air-speed and to some extent by the attitude of the aircraft relative to its trajectory. As for the acoustically relevant properties of the air, the same considerations apply as in the previous subsection.

When the engines are used as auxiliary brakes after landing, i. e., when they are in reverse thrust mode, the thrust generating airflow is redirected contrary to the direction of the aircraft movement. The exhaust air of the engines thus flows toward the flight deck. This causes additional noise.

The engine sound propagates partly through the structure of the aircraft, partly through the air outside. Due to the reflection off the runway, this latter propagation path is more distinct when the aircraft is on ground.

In the aircraft we are modeling there are two jet-engines, each with two internal rotors. Thus, we deal with a total of four rotors not generally revolving at exactly the same frequency. This lack of permanent complete synchronism causes characteristic beatings in the sound (also known as fluctuation strength).

The engine noise is reflected in Figure 2 by some of the distinct peaks, namely the ones at $f \approx 190$ Hz and multiples thereof.

2.1.3. *Sound of Rolling Aircraft Wheels*

When the aircraft rolls on the runway during take-off and landing phases, a structure-borne sound is produced, the cause of which is the friction between the tires of the landing gears and the ground, in combination with the unevenness of the runway on different scales (cf. Figure 3).

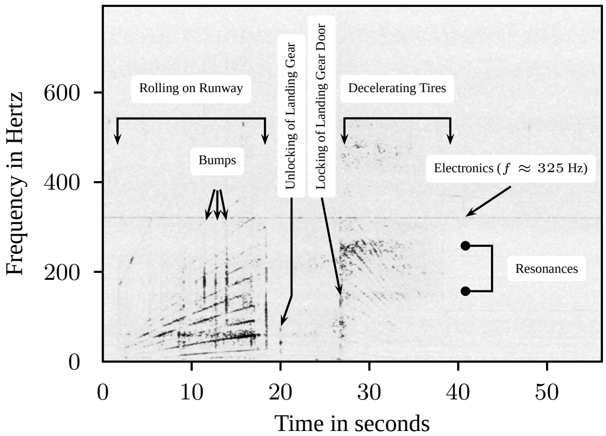

Figure 3: Spectrogram, around the time of take-off. See various subsections for details.

The quality of the sound of rolling aircraft wheels depends on the relationship between rotatory speed of the wheels and sliding speed of the aircraft, as well as on the contact pressure of the tires on the ground. Furthermore, the sound is influenced by the condition of the tires (profile), the brake system (e. g. ABS), and the suspension of the landing gear. Finally, water, snow, or ice present on the ground are determinant factors as well.

2.1.4. Sound of APU

The auxiliary power unit (APU) is a source of energy located at the tail of the aircraft. During operation, it emits a loud noisy sound. Like the engine noise, the sound of the APU audible on the flight deck is partly structure-borne and partly airborne.

Before engine start, the APU can be switched on independently of other on-board systems. In contrast to most other acoustic events, its sound can therefore be recorded separately.

2.1.5. Sound of Rain on Front Windshield

When an aircraft flies through a rain shower, a clearly audible, pattering sound is produced by the rain drops. The heavier the rain, the louder the sound. Furthermore, the quality of the sound depends on the cruising speed and the attitude of the aircraft. Clearly, the direction of arrival of the corresponding sound is more or less the whole front part of the aircraft, in particular the front windshield.

2.1.6. Sound of Air Condition and Electronics

On the flight deck, there is a contribution to the sound field from electronic equipment such as inverters and avionics. The buzzing sound is responsible for the distinct peaks in the Power Spectral Density shown in Figure 2. See also the continuous line at the frequency $f = 325$ Hz in Figure 3.

2.1.7. Sound of Moving Parts

The moving parts on the outside of an aircraft—flaps, landing gears, and speed brakes—are positioned by servo motors. These motors do produce sounds, which differ for various kinds of motors, e. g. hydraulic vs. electrical.

The sounds in this class can be considered temporary, i. e. they persist only as long as the corresponding part is in motion.

Our aircraft being large, basically the sounds of the moving parts located in the rear are almost inaudible. Therefore, we can neglect potential sounds from the flaps and the speed brakes.

2.2. Discrete Sounds

Unlike the continuous sounds described in the previous section, certain sounds occur in consequence of specific events. Approximately, these "discrete" sounds do not depend on any parameter, are reproduced when needed, and this reproduction cannot be stopped prematurely. The sounds in this class include:

2.2.1. Bumps on Runway

In addition to the sound of the rolling aircraft wheels described in Section 2.1.3, there is a squealing sound at the moment the aircraft touches down. We can also observe distinct bumps whenever the aircraft moves over the joints between concrete slabs on the runway. An example of such bumps can be seen in Figure 3.

2.2.2. Sound of Decelerating Tires

Consider the phase immediately after take-off, when the landing gears are already retracted. According to our information, the tires are then decelerated by an external force. A harmonic sound with falling fundamental frequency results. For the time being, we suppose that the rate of this decay is approximately constant. However, the wheel well of the landing gears causes pronounced resonances with time-invariant center frequencies and bandwidths. Both effects are clearly visible in the spectrogram in Figure 3.

2.2.3. Sounds of Locking Parts

There are often additional sounds when a part starts moving respectively when moving parts lock into their final positions (e. g. landing gears). Again, an example can be found in Figure 3.

3. DATA ACQUISITION

All necessary recordings were made in the cockpit of an aircraft during several scheduled flights. Consequently, as was inevitable, the recordings are deteriorated by the pilots' interfering communication in the cockpit. Even so, since the quality and the quantity of the acquired data sufficed for our purposes, we did not have to use noise-reduction techniques. Instead, we relied on the "clean" segments as representative.

In order to obtain a most thorough acoustic picture of the cockpit, we used two recording techniques in parallel, as described in the rest of this section.

3.1. Binaural Recording Technique

The first recording technique comprised a high-quality binaural microphone system by Brüel & Kjær. The small and lightweight system allowed for extremely comfortable mounting and use. Furthermore, it features a full-range frequency response and can be calibrated, thus providing the basis for objective measurements. In

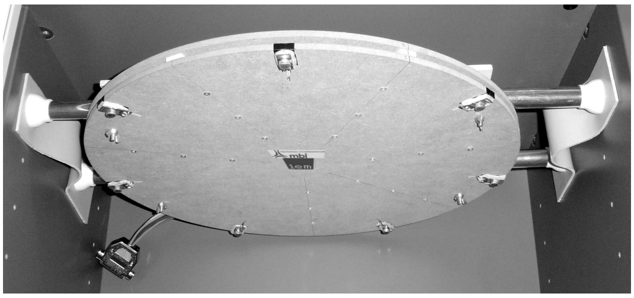

Figure 4: Microphone array mounted in a niche. The setup is similar to the one on the flight deck.

addition, the binaural signals are well suited for a subjective judgment about where in three-dimensional space the particular sounds come from.

On the other hand, the recordings made with the binaural microphone system are affected by whatever head movements the person wearing it—the pilot—has made during recording, which has to be considered during signal analysis.

3.2. Multi-Channel Recording Technique

As a second recording technique, we employ a planar array of microphones. Our prototype is shown in Figure 4. As a beamformer, such an array is the basis for spatial sampling. More specifically, it can be used to computationally assess the direction of arrival of the different sounds.

The cockpit geometry requires that the microphone array be positioned behind the pilot's seat in a niche[1] about half a meter in diameter. Consequently, we chose the diameter d of our prototype array as $d = 0.45$ m. Based on our recording equipment, we can use a total of $N = 7$ microphone channels.

The frequency range of the low-cost microphones themselves spans $250\,\text{Hz} < f < 6\,\text{kHz}$ at -6 dB. Such a limited range is justifiable because at frequencies below 380 Hz, beamforming in the sense of a delay-weight-sum (DWS) beamformer [4] is not feasible anyway given the array aperture, and because the power density spectrum of the sound pressure level decays toward higher frequencies (Figure 2).

4. SIGNAL ANALYSIS

In this section, we describe the methods and techniques applied to the analysis of the recorded data for each of the acoustic events discussed previously.

Additionally, we performed simple inspection of the waveform and of the spectral structure of a time-signal when needed. For example, in the case of the sound of decelerating tires, we determined both center frequency and bandwidth of the resonance directly from a spectrum estimate. Finally, mechanical data recorded simultaneously during the flight sometimes helps, e. g. the rotary speed of the engines is known and can be used in the analysis of the engine noise, and likewise for the sound of the aircraft rolling on the runway.

[1] Unfortunately, this causes unrecoverable alterations in the sound, e. g. standing waves.

4.1. Linear Prediction Analysis (Noise Due to Air-Current)

On an experimental basis, we discovered that the overall acoustic situation in the cockpit as recorded by the binaural system may appropriately be reproduced by some kind of filtered noise. Then, the dependence of the sound on certain parameters (Section 2) becomes manifest in different power spectral densities (Figure 2). In other words, the different flight states can be described by different filter curves.

In order to parameterize the different flight states, we use the technique of linear prediction analysis [5]. This technique models a signal $x[n]$ by

$$x[n] = \sum_{j=1}^{P} a_j[n]x[n-j] + \sigma_\nu^2 \nu[n], \qquad (1)$$

where $\{a_j[n]\}_{j=1}^{P}$ are IIR filter coefficients at discrete time n and $\nu[n]$ denotes samples drawn from a white-noise process with variance σ_ν^2. As for the probability density function of $\nu[n]$, an analysis of prediction-error signals corresponding to several different flight situations showed that the random variable $\nu[n]$ is best modeled as Gaussian distributed.

The representation according to Eq. (1) provides a very compact, small-set parameterization of the sound in a given flight state. For example, when modeling 31 poles, we get only $31 \cdot 2 + 1 = 63$ parameters per flight state.

In order to provide a higher degree of realism, a few sinusoidal components should be added to the filtered noise. This is further discussed in Sections 4.2 and 5.3.

4.2. Peak Picking (Air Condition and Electronics)

The distinct peaks in the spectrum (Figure 2) can be attributed to the sounds from electronics and the air condition. Their exact location can be found by a standard peak-picking algorithm fed with the prediction-error signal [5] from the linear prediction analysis, with the contribution from the engine noise subtracted. Moreover, since the positions of the peaks stay relatively constant over time, temporal information can be incorporated in the peak detection.

In general, only ten to twenty peaks lie above the masking threshold.

4.3. Matching Pursuit Algorithm (Bumps)

The extraction of the bump sounds on the runway was accomplished with a matching pursuit (MP) algorithm [6, 7]. Compared to methods such as the Digital Fourier Transform (DFT), the MP algorithm provides a better trade-off between time resolution and frequency resolution [7]. It does not assume a harmonic structure in the signal. Instead, damped exponential sequences are used as the signal kernels [6]:

$$x[n] = \sum_{k=1}^{Q} \rho_k S_k \underbrace{r_k^{n-\Delta_k} \mathrm{e}^{\mathrm{j}\omega_k(n-\Delta_k)} u[n-\Delta_k]}_{\substack{\text{damped} \\ \text{exponential}}}. \qquad (2)$$

Here, ω_k, r_k, and ρ_k are the frequency, the damping constant, and the amplitude of the damped exponential sequences, respectively. S_k is a scaling constant. Δ_k denotes a time shift and $u[n]$ the unit step sequence so that the kth exponential sequence is applied at time Δ_k.

Extracting the bump sound by means of the MP algorithm with damped exponential sequences is adequate because as a kind of impulse response, the sound resembles an exponentially decaying sequence. Moreover, the MP algorithm is well suited for the analysis of transients since it uses asymmetric atoms [6].

Based on the MP analysis, a time signal of the "clean" bump sound can be computed and stored for later playback.

4.4. Linear Filtering, Notch Filtering (Sound of Rain)

For the isolation of the sound of the pattering rain we used a combination of gentle high-pass filtering and notch filtering. More exactly, a comparison between the power spectral densities of segments with rain respectively without rain revealed that the energy of the sound of rain is concentrated in the region of higher frequencies. On the other hand, we tuned notch filters manually at the harmonics of the fundamental frequency corresponding to the instantaneous rotary speed so as to eliminate the engine noise.

In conclusion we may say that this simple procedure worked surprisingly well.

4.5. Beamforming (Localizing Sounds)

We plan to use a combination of a constrained robust near-field beamformer with a conventional DWS beamformer in order to localize our sounds. For example, if for a known sound, most of the energy comes from a certain direction, the source of that sound will be mapped to this direction.

5. SIGNAL SYNTHESIS AND SPATIALIZATION

The synthesis techniques complementary to the analysis are presented in this section.

5.1. Linear Prediction Synthesis

When synthesizing the noise due to air-current, we implement the IIR filtering in Eq. (1) by an all-pole lattice structure, parameterized by reflection coefficients [5].

The all-pole lattice filter implementation is more convenient in terms of stability in the bounded-input, bounded-output (BIBO) sense when transitions between two different flight states need to be computed. More specifically, imagine that at time $n = 0$ our lattice filter is in state $\{\kappa_j[0]\}_{j=1}^{P}$. Then, a smooth transition to the set of new reflection coefficients $\left\{\kappa_j^{\text{new}}\right\}_{j=1}^{P}$ corresponding to a new flight state can be obtained by the first-order recursion

$$\kappa_j[n+1] = (1-\gamma)\,\kappa_j^{\text{new}} + \gamma\,\kappa_j[n], \qquad 1 \le j \le P, \tag{3}$$

where the constant $\gamma < 1$ determines the rate at which the new set of reflection coefficients $\left\{\kappa_j^{\text{new}}\right\}_{j=1}^{P}$ is reached. If the absolute value of all coefficients from both sets is bounded by unity, the filter described by the coefficients $\kappa_j[n]$ in Eq. (3) is always BIBO-stable, based on a property of reflection coefficients [5].

5.2. Looping – Wavetable Readout

As soon as we have succeeded in isolating sounds such as bumps on the runway, the pattering sound of rain during the flight, or the sounds of moving parts, synthesis is easy for this class of sounds: we simply loop the corresponding takes with constant read-out speed (bumps) or with variable read-out speed (rain), whereas the sounds of moving parts are reproduced playing back a wavetable just once.

In case the transition of a moving part stops for any reason, the wavetable playback is interrupted at once. Then, however, further sounds can emerge. For example, when the nose landing gear is blocked without reaching its final position, the landing gear door cannot close, and the resulting cavity produces an additional noisy sound (cf. Section 2.1.1).

5.3. Additive Synthesis

As for the sounds produced by electronics and air condition, simple additive synthesis of few dozens sinusoidal signals with the correct amplitudes and frequencies is performed.

5.4. FM-Synthesis

The harmonic spectra of sounds like engine noise, the sound of the rolling aircraft wheels, and the sound of decelerating tires can easily be synthesized using frequency modulation (FM), with both modulation frequency and carrier frequency tuned to match the rotary speed of the corresponding moving part. The modulation index is adjusted depending on the bandwidth of the desired spectrum. Subsequent resonance filtering might be necessary.

5.5. Peak Filtering

The sound of decelerating tires is re-synthesized by filtering the signal from an FM-synthesizer with a second-order band-pass filter with adjustable center frequency and bandwidth.

5.6. Spatialization

From a signal processing point of view, the generator modules incorporated in our sound system are all single output systems. Obviously, in order to provide the required degree of realism, these monophonic signals need to be spatialized properly. Depending on the type of a specific sound, different spatialization strategies are pursued.

5.6.1. Positioning of Localizable Sounds

In principle, from the pilot's seat some of the sounds described in Section 2 should be localizable quite well, e. g. the sounds emanating from the APU, the engines, or the landing gears. Ideally, such sounds are best reproduced by a loudspeaker located at their exact physical position. But in the cockpit of the simulator we have a limited number of loudspeakers at predetermined positions not generally matching the desired ones (cf. Section 1.2).

However, we can do "virtual sound source positioning" using vector-base amplitude panning (VBAP), a technique proposed by Ville Pulkki that is computationally efficient for static sound sources [8]. With this technique, static virtual sources can be positioned anywhere within the spherical triangles spanned by loudspeaker triples.

5.6.2. Providing Spatial Extent

In contrast to the situation described in the previous subsection, sounds such as air-current noise do not seem to originate from a concisely defined point source. Instead, they apparently possess spatial extent.

Starting again from a single-channel signal, such spatial extent can be rendered by decorrelation techniques like those proposed in [9]. If, for example, two loudspeakers are fed by uncorrelated noise signals, the noise appears to fill the complete space between them [10].

In a similar fashion, k different decorrelated versions $d^{(k)}[n]$ of the monophonic signal $s[n]$ to be spatialized can be obtained from linear filtering:

$$d^{(k)}[n] = s[n] * a^{(k)}[n] \qquad \forall k. \tag{4}$$

Here, the symbol $*$ denotes discrete-time convolution and $a^{(k)}[n]$ is the impulse-response of a filter with flat magnitude response but random phase response, i. e. an all-pass filter [9]. In our case, k equals the number of loudspeakers in the setup, i. e. we employ one filter per channel.

The procedure presented in the previous paragraph accomplishes decorrelation of a signal over the whole frequency range. In certain circumstances though, it might be better to provide a source with different degrees of spatial extent in various frequency bands. More precisely, consider splitting a monophonic source signal $s[n]$ in a high-pass channel $s_{\mathrm{HP}}[n]$ and a low-pass channel $s_{\mathrm{LP}}[n]$. Then—from a physical point of view—it might be clever not to decorrelate the low-pass channel $s_{\mathrm{LP}}[n]$, since the spatial distribution of sound pressure level is approximately constant at low frequencies.

6. CONCLUSION AND FUTURE WORK

In this paper, we presented a sound system for a realistic, modular flight simulator. We showed that viable audio material can be obtained from a combination of binaural and multi-channel recordings, because then spatial information, too, can be assessed. We discussed both continuous and discrete acoustic events typically occurring on the flight deck of an aircraft and their dependencies on physical parameters. We proposed various techniques for the extraction of these acoustic events from real data. Above all, linear prediction analysis has proven useful for this purpose. Finally, issues of spatialization were addressed.

Up to now, in our project we have mainly dealt with recording respectively system-design issues. Additionally, we have assembled a rudimentary prototype (Figure 5), where in the presence of actual pilots we have proven the feasibility of our concepts. In future work we will focus on analysis and signal processing techniques, e. g. beamforming and blind deconvolution [5]. Moreover, the matching pursuit algorithm could be a helpful tool not only for the extraction of bumps, but for other sounds as well.

7. ACKNOWLEDGEMENT

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8. REFERENCES

[1] "Aeroplane flight simulators (amendment 3)," http://www.jaa.nl/section1/jarsec1.html, 2005.

[2] Siegfried Vössner, Reinhard Braunstingl, Helmuth Ploner-Bernard, and Alois Sontacchi, "A new framework for a sound system for realtime flight simulation," submitted to DAFx'05, Madrid, Spain, September 2005.

[3] Pure Data web site, "http://pd.iem.at," March 2005.

[4] Michael Brandstein and Darren Ward, Eds., *Microphone Arrays*, Springer, 2001.

[5] Simon Haykin, *Adaptive Filter Theory*, Prentice Hall, 4th edition, 2002.

[6] Michael Goodwin, "Matching pursuit with damped sinusoids," in *Proc. ICASSP '97*, Munich, Germany, April 1997, vol. 3, pp. 2037–2040.

[7] Garry Kling and Curtis Roads, "Audio analysis, visualization, and transformation with the matching pursuit algorithm," *Proceedings of the 7th International Conference on Digital Audio Effects (DAFx'04)*, pp. 33–37, October 2004.

[8] Ville Pulkki, "Virtual sound source positioning using vector base amplitude panning," *J. Audio Eng. Soc.*, vol. 45, no. 6, pp. 456–466, June 1997.

[9] Guillaume Potard and Ian Burnett, "Decorrelation techniques for the rendering of apparent sound source width in 3d audio displays," in *Proceedings of the 7th International Conference on Digital Audio Effects (DAFx'04)*, Naples, Italy, October 2004, DAFx, pp. 280–284.

[10] Gary S. Kendall, "The decorrelation of audio signals and its impact on spatial imagery," *Computer Music Journal*, vol. 19, no. 4, pp. 71–87, 1995.

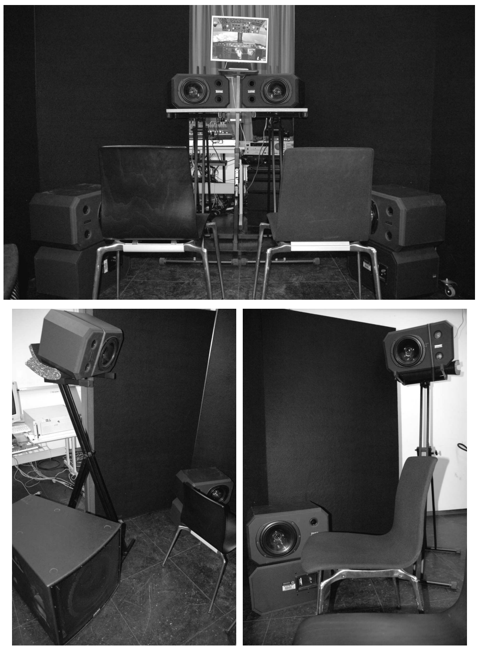

Figure 5: Prototype of sound system