An evaluation of audio systems of the BioMag laboratory

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1. Introduction

We studied acoustics and audio systems in the BioMag laboratory at the Helsinki University central hospital (December, 2003). BioMag is a brain research laboratory for non-invasive measurements of human brain based on magneto- and electroencephalography, MEG and EEG respectively. Both EEG and MEG devices are sensitive to electro-magnetic interference. Isolating magnetic interference from MEG measurement is technologically challenging. For the MEG device it is necessary to build a magnetically shielded room which is a massive and costly construction. Moreover, MEG is sensitive to any electrical device brought in its vicinity inside the shielded room.

A large part of the scientific work in BioMag relates to the auditory brain research. However, the auditory stimulation in the measurements is not a straightforward issue, because the sensitivity of the MEG device excludes the use of any conventional loudspeakers, headphones or, in fact any electroacoustic transducer. Therefore, in practice the sound stimuli is lead to the listener’s ears through plastic tubes via an insert earpiece. Design of such audio systems are discussed in the studies by Airas et al. (1999) and Riederer et al. (2002). The goal of this study was to investigate the performance of the two audio systems used in the BioMag. In addition we investigated background noise levels of the devices, which is a commonly known problem in the BioMag.

The responses of the audio systems were measured using swept sinusoid technique allowing simultaneous recording of the linear and non-linear system response (Müller & Massarani, 2001; Farina, 2000). The paper is organized as follows: first Section 2 describes the measurement methodology and the equipment. Secondly, in Section 3, the results of the measurements on the NeuroMag audio system are presented. This section is organized so that the basic frequency response, group delay and background noise measurements are shown first. Then as the system can be used with different lengths of the tubes the effect of different joint pieces is demonstrated. Further, we investigate the behaviour of the sound field in the different parts of the tubes. Further we evaluate the distortion of the system and frequency responses of the various distortion orders. In the Section 4 the custom made wideband audio system is investigated. First we show basic measurements of the frequency responses, group delays, background noise and crosstalk. Secondly, we investigated the random effect of the placement of the ear insert plugs.

2. Measurement methods

The measurements were performed using a laptop equipped with a high quality sound card. Matlab based software was used for generation of the excitation. The audio tubes are attached to the listener’s ears using an ear insert piece made of a foam plastic ear plug. The ear insert piece is attached to the tube using a thin pipe that transmits sound through the foam plastic piece. Acoustic signals from the tubes were recorded using Bruel & Kjaer artificial ear simulator 4157, including a model of ear canal and a B&K condenser microphone 4143. The ear simulator was connected to B&K microphone amplifier which was
further connected to the preamplifier and the PC sound card.

The impulse response measurements were performed using a logarithmic sweep excitation signal generated in the frequency domain (e.g. Müller and Massarani, 2001; Farina, 2000). The sampling rate used was 48kHz. The sweep started at 50 Hz and ended at 20 kHz frequency. The actual measurement signal consisted of sweep (90% of the duration) and a silence period (10% of the duration). The length of the whole signal was $2^{16}$ samples corresponding to a duration of 1.365 sec. In order to increase signal to noise ratio (SNR) sweeps were repeated 8 times in each measurement over which the responses were averaged. A rectangular time window was used in the deconvolution.

3 NeuroMag audio system

3.1 Basics

In this section we present the result from the measurement of NeuroMag acoustic tubephone audio device. It consists of a small piezo electric driver from which the acoustic signal is further transmitted through a thin plastic tube to the listener’s ear. The different length of tubes can be used by attaching 26 cm joint pieces. Impulse responses were measured, firstly, for the tubes of the both channels using two joint pieces, which is the setup used normally in brain experiments. Secondly, we measured impulse responses for the tubes with one, two or three joint pieces.

The impulse response for both channels with two joint pieces are plotted in Figure 3.1. The response is normalized so that the largest peak is at 1 dB. Hence, the SNR of the measurement is around 95 dB, which can be seen directly at the background noise levels after the decay of the response. The frequency response of the channels one and two with two joint pieces are shown in Figure 3.2 A and B, respectively. The tubes have passbands (indexed by 3dB points of attenuation) between 70 Hz and 4 kHz. The frequency responses at the passband deviate maximally 6.5 dBs at channel one and 8 dBs at channel two. Figure 3.2 C shows the comparison of the two tubes channels (with two joint pieces) obtained by subtracting the response of channel one from channel two. Channel one is from 0 to about 5 dBs louder within the passband. Figure 3.2 C shows also that the differences in the group delay between the two channels are within 0.5 ms at the passband.
Figure 3.2 Frequency responses and group delays for short pipes for the stereo channels (A) one, (B) two and (C) the differences between two channels obtained by subtraction of (A) and (B). (FFT length $2^{14}$).

Figure 3.3 demonstrates the effect of the tube length to the frequency response and group delay. With the shortest tube length frequencies from 1kHz to 3kHz are amplified comparing to response amplitude below 1kHz. All the responses show a spurious peak around 10 kHz. However, the amplitude of this peak is reduced along the increasing tube length as the high frequency part attenuates. The effect of the tube length is demonstrated perhaps most clearly in Figure 3.3 D in which each response is normalized by subtracting the response of the shortest tube. Each tube has a convolutional effect with increasing attenuation towards high frequencies. Small amplitude ripple in the responses originate from the reflections of the speaker and microphone terminations of the tubes. Both ends produce reflections due to the changing acoustic impedance. Increasing pipe length can be seen as constant shift across frequencies in the group delay, which is proportional to the tube length. In theory identical joint pieces of the tubes will have constant attenuation to the acoustic field which is seen additive in log spectral domain. In the next section we will derive the attenuation of joint pieces more precisely.
In addition to the audio system responses another relevant factor is the background noise during the auditory experiment. Particularly interesting is the noise level entering subject’s ears during the experiment rather than noise levels observed in the room. Part of the noise observed in the room will be blocked by the ear insert plug. Noise can be conducted to the tubes through speaker boxes or through the tubes’ plastic walls. Figure 3.4 shows the noise spectra for most commonly used tube length two. Total A-weighted equivalent noise level summed over audio frequency range $L_{aeq} = 27$ dB is obtained. For summing of levels in dB see Lahti (1997) formula 1.33 in page 18. For frequency weighting and SPL measurement see Lahti (1997) chapters 1 and 2.

Figure 3.3 Frequency response and group delay for different lengths of tubes. (A) one piece (B) two joint pieces (C) three joint pieces. (D) All the frequency responses normalized by subtracting the response (A). More attenuation towards high frequencies is observed with the increasing tube length
3.2 Distortion

Swept sine technique makes it possible to measure the linear response impulse response and distortion of the system simultaneously (Farina, 2000). During the excitation the distortion components will appear as multiples of the instantaneous frequency of the sweep. The system response can be obtained by removing the excitation from the recorded signal by deconvolution. Distortions will appear well separated before the linear response of the system, in which each distortion order will have its own impulse response. For the logarithmic sweep time lag of the distortions from the linear response \( \Delta t \) can be calculated precisely using the following relation (Farina, 2000),

\[
\Delta t = T \frac{\ln(N)}{\ln\left(\frac{\omega_2}{\omega_1}\right)}
\]

where \( T \) is the duration of the sweep, \( N \) is distortion order and \( \omega_1 \) is the start and \( \omega_2 \) is the end frequency of the sweep. As seen from the formula, the placement is a logarithmic function of the distortion order. This means that longer durations of impulse response of the low order distortions can be extracted, whereas higher order distortions appear as packed denser. In theory high order distortions will continue throughout the linear phase of the response. In practise, however, their amplitudes are very small comparing to SNR. By increasing the duration of the sweep the space in time for each of the distortion orders can be increased.

Responses are shown for the two systems, first with pipe length one (Figure 3.5, left hand plots) and the second with tube length two (right hand plots). The distortion components can be seen in the end of the time domain response. Note that the response in theory is a periodic function and this figure shows only one period of it, therefore the distortion components seen in the end of the response can be interpreted as occurring before the linear response. The first distortion order starts from the last transient seen in the plot.

Magnitude spectra plots in Figure 3.5, from top to bottom, show first the linear response of the system and then the spectra of selected distortion orders. Note that in this case the distortion components correspond to output distortion, in which the distortion of the loudspeaker and audio system is convolved with the linear response of the system. When quality of auditory stimulation with this device is considered, total output distortions are more interesting than original distortion generated by the speaker. Probably, the most remarkable source of distortion is the piezo electric speaker. If the distortion generated by the speaker is needed to be accessed directly, it can be obtained by removing the response of the linear system by deconvolution (or subtraction in the log-spectral domain).
Figure 3.5 Analysis of the output distortion. Results for the tube length one (left panel) and two (right panel) are shown. Order of distortion components is marked at the corresponding peaks of the response, and at the top left corner of each spectral plot.
4. Wide band audio system

4.1 Basics

A wideband audio system of the BioMag laboratory was also measured. Time domain system responses of both the channels are shown in Figure 4.1. The response is normalized so that the largest peak is at 1 dB. The first transient in the plot represents the response of the linear phase of the system and the transients at the end correspond to distortions. Distortions appear to be 40 dBs below the linear response. The SNR of the measurement (around 95 dB) can be seen from the background noise level after the linear response. The passband of the system is initially designed up to 11 kHz. Figures 4.2 A and B show the frequency responses and group delays of the channels one and two. The deviation in the frequency responses above 3 kHz are rather large (maximally 20 dB from peak to peak, in the channel two). In contrast, the deviations in the group delay are largest at low frequencies below 400 Hz. The deviations are probably caused by the reflections from the ends of the tubes. Figures 4.2 C and D show a comparison of the two channels obtained by subtraction. Figure 4.2 C show non-smoothed magnitude spectrum and Figure 4.2 D 1/3 octave smoothed version. The 1/3 octave smoothing is applied since it corresponds well to the auditory filter bandwidth (Moore and Glasberg, 1984). Passband SPL deviations between the channels are largest in the high frequency range above 4 kHz being maximally 24 dB from peak to peak. In contrast the maximal deviations in the group delay occur at low frequency range below 500 Hz. In the smoothed curves the deviations are greatly reduced. As a comparison, Figure 4.2 E shows the original response of one channel of the tubesystem, which was measured when the tubes were new.

Background noise is known to be a problem with the wide band audio system. Part of the noise originates from the noisy room where the speakers are placed and part is of digital origin for the DSP equalization device. In order to investigate the background noise we measured the A-weighted equivalent noise spectral levels measured from both tubes. The magnitude spectra are plotted in the Figure 4.3. When summed (across frequencies a total A-weighted noise level of $L_{a_{eq}} = 27$ dB, was obtained. For summing of levels expressed in dBs see Lahti (1997) formula 1.33 on page 18. For frequency weightings in SPL measurement see Lahti (1997) chapters 1 and 2.
Figure 4.2 Frequency responses and group delays measured for the wide band audio system channel (A) one and (B) two. Differences in the frequency response and group delays between the channels by subtraction using (C) linear resolution spectra and (D) 1/3 octave smoothed spectra. (E) For a comparison a measurement from original design phase of the tubes is shown.
In order to investigate whether the sound is conducted from one channel to another we investigated the cross talk by exciting channel one and measuring the response from channel two, when the tube opening of channel one was blocked. Figure 4.4 shows spectra of measured crosstalk and the spectra of crosstalk subtracted from the linear response. It appears that crosstalk is 100 dBs below the response in the passband which is already at the level of background noise of the measurement.

4.2 The effect of ear tip replacement

The acoustic tubes are attached to the subject’s ears by an ear insert piece made of foam plastic ear plug, which is attached to audio system by a very thin tube inserted in the plug. Attaching these ear plugs to the listener’s ear evidently is a random factor that might have different effect to tube responses depending how it is placed. We investigated this effect measuring four ear insert pieces. Figure 4.5 A shows the magnitude spectra of these four measurements. Figure 4.5 B shows responses normalized by subtracting the magnitude spectra for the first ear tip (Figure 4.5 A, top-panel). In the normalized plots the deviations are the largest at high frequency range above 7kHz, in which maximal deviation (peak to peak) is 20 dBs. Figure 4.5 D shows standard deviation over the four measurements, which is maximally 4.5 dB at
5. Discussion

In the present study we measured the audio systems of the BioMag brain research laboratory. Both original NeuroMag audio and the custom made wideband audio systems were investigated. A new technique using swept sinusoids was applied for simultaneous recording of impulse response and distortion (Farina, 2000; Müller et al., 2001).

Construction of an MEG compatible audio device is a rather special design task, since the magnetic sensitivity of MEG excludes the use of traditional headphones or speakers. Common solution has been to place the electric parts far from device and lead the sound to subject’s ears via plastic tubes. In the BioMag there are two different variants of this type of design. The original NeuroMag’s solution was to use a small piezo speaker, which can be driven by so small currents that it can be placed in to the magnetically shielded measurement room in the distance of few tens of centimetres from the MEG device. Another solution is to use larger conventional dynamic speakers which can only placed outside the measurement room. In this case the sound is led to longer tubes which is further led through the wall of the high frequency range. Hatched lines indicate the uncertainty of the estimate which is inversely proportional to square root of the sample size (four).
magnetically shielded room. Those tubes are thicker near the speaker and are narrower near the listeners ears. Thicker tubes have less damping especially at high frequencies than narrower. Therefore, it has been possible to have wider frequency band using thick tubes. However, no industrial manufacturer exist for such system, and therefore the tubes are custom made. The second device also involves a DSP equalization device, which attempts to correct the lowpass spectral shape of the sound lead through the tubes, and the ripple at the passband caused by the echoes from the ends of the tubes. The DSP device implements an inverse filter for the response measured from the tubes using an ear simulator. Initially, it has been possible to equalize the tubes up to 11kHz. Currently, however, the frequency responses of the tubes are not flat at that range. These changes probably originate from replacement of some broken parts in the joints of plastic tubes and replacement of the speakers with ones that are not exactly the of the same type as the originals. Moreover, the joint piece of ear tip and tubes seemed to be slightly damaged causing probably some acoustical leakage. In addition, the length of the tubes may not have changed slightly as the joints seem not to be fixed at present. This has particularly damaging effect to the cancellation of the echoes from the ends of the tubes. In order to remove effects of the echoes accurate timing in the designed correction filters is required. It is for emphasising that in this measurement series we measured only DSP equalized responses. Measurements of the system without equalization, and the measurements of the DSP device alone were omitted.

Another problem of the wide band audio system is relatively high background noise level (40 and 41 dB) due to unsuitable placement of the speakers and the background noise of the DSP equalizer. Currently the speakers are placed in a metal shelf in noisy room, including MEG cooling system machinery and computers, without any acoustical insulation. From that room noise is transmitted acoustically through the speaker boxes and tubes to the listeners ears. Similar noise problems are not observed with the NeuroMag audio system, as it is placed inside the shielded room which is not as noisy. In addition, it does not have any digital equalization that might not increase noise levels. More thorough examination of the real source of noise would have demanded measuring separately the noise by to audio equalization and noise conducted from the machinery room through tubes.

The analyses in replacement error of the ear tip revealed rather large errors in the high frequency part of the tube responses. In fact these errors seem to be of similar magnitude as the differences between two channels. Thus at least partly the differences between the channels might originate from different placement of the ear tip.

For the NeuroMag audio system we measured distortion and analyzed magnitude spectra of some distortion orders. Distortion for tube length one was of much larger amplitude. However, part of this can be explained by more high pass character using tube length one when comparing to tube length two.

As a conclusion the NeuroMag audio system meets fairly closely the specifications for which it has been designed. However, wide band audio would need thorough repairment including re-estimation of DSP equalization filter, changing damaged parts in the tubes and improving the acoustic insulation.

Acknowledgement

This is a part of a course work of “S-89.430 Akustinen mittaustekniikka” in the Laboratory of Acoustics and Audio Signal Processing in the Helsinki University of Technology. The author wishes to thank Mr. the teacher of the course Timo Peltonen (Akukon Oy, Kornetintie 4 A, 00380 Helsinki), who was the main responsible of collecting the acoustical measurement data, and who provided the measurement equiipment.

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