# The MarkIII microphone array: the modified version realized at ITC-irst

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## 1 INTRODUCTION

The objective of this document[1] is twofold.
Firstly, we address the various problems so far experienced at ITC-irst for what concerns the use of the Mark III microphone array [1]. The problems are grouped in the following seven categories:

1) Early saturation effect of microphones
2) Packet loss and other inconsistencies
3) 50 Hz disturbance
4) Device noise and one-sample delay
5) 8 and 16 kHz common ground noise
6) Anomalous 22 kHz behavior
7) Potential microboard breakdown

In the next section, each problem is addressed in more detail. Note that the first three points are reported (although they could be considered as trivial) because some partners in CHIL [2] could find useful verifying their entity in the recordings so far done at their site. On the other hand, problems 4-6 refer to very crucial aspects, which may cause misleading results in source localization, beamforming, or any other multi-channel processing based on the assumption of electrical noise independence across channels.
In some cases, the tools used to evidence the given problems are not described in detail, assuming the readers are familiar with the related concepts (as, for instance, cross-power spectrum phase analysis). Some of these tools will be made available soon in the CHIL web site. References are anyway reported for a further investigation and we are always available for further clarification.

The second objective of the document is to provide an expert in electronics with enough

[1] This second draft of the document adds more information to the previous version for what concerns details for the reproduction of the modified Mark III array. The final version of the document will be completed once we will have checked the convenience of some further modifications at this moment under study.

information to reproduce a modified sample of the device. To this purpose, Section 3 reports on the way each problem was solved, while in Appendix A we report the component list necessary to the modification of the array.

Finally, Appendix B will provide some pictures showing various details of the resulting device.

# 2 DESCRIPTION

## 2.1 Early saturation effect of microphones

It was observed that when a speaker is near the array the microphone signals immediately saturate. We guessed that the Panasonic microphones were too sensitive or the OPAMPs were pushed to the limit.
In any case, the device does not allow to control input levels.
Moreover, it is worth noting that some microphones are more sensitive than others. The biggest ratio from the most sensitive (ch 35 and ch 8, respectively, in the array available at IRST) was of 2:1, i.e. 6 dB in amplitude.

## 2.2 Packet loss and other inconsistencies

We experienced a packet loss collecting data both at 44.1 kHz and 22.05 kHz, even with very short recordings.
Note that with a 100 Mbps Ethernet connection based on a dedicated switch between the array and the PC, a 22.05 kHz acquisition is affected by some packet losses, while a 44.1 kHz acquisition experiences even seconds of packet losses, corresponding roughly to more than 4000 Ethernet packets lost in one second.
We also observed in the very first far-microphone recordings done at UKA (seminar of July, 21st 2004) that several packets were lost, although they had been recorded at 22.05 kHz.
Other inconsistencies were also observed for what regards the beginning of any recorded signal. Broadly speaking, the first 4350 samples (at most) can include either a sequence of zero samples or a short sequence of random samples or a spike at the very beginning.
Moreover, the first samples are characterized by an increase of the signal from a negative level, which should correspond to a charging process in a capacitor (see Figure 1).

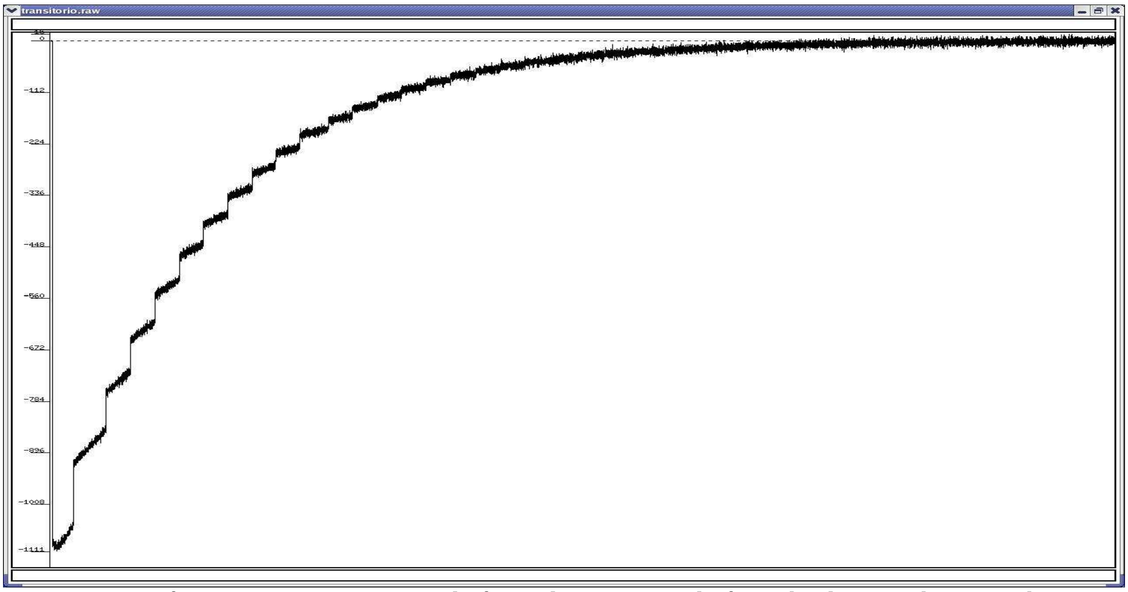

Figure 1: Waveform containing one second of signal coming purely from the devices: the microphone was in fact isolated. A transitory is well visible, due to capacitors charge, as well as the device noise. This suggests to throw out the first 4350 samples at 44100 Hz, i.e. the first 100 ms of each recording.

### 2.3 50 Hz disturbance

In our preliminary recordings (done in a very quiet and acoustically insulated room) we had observed the presence of a perceivable 50 Hz interference.

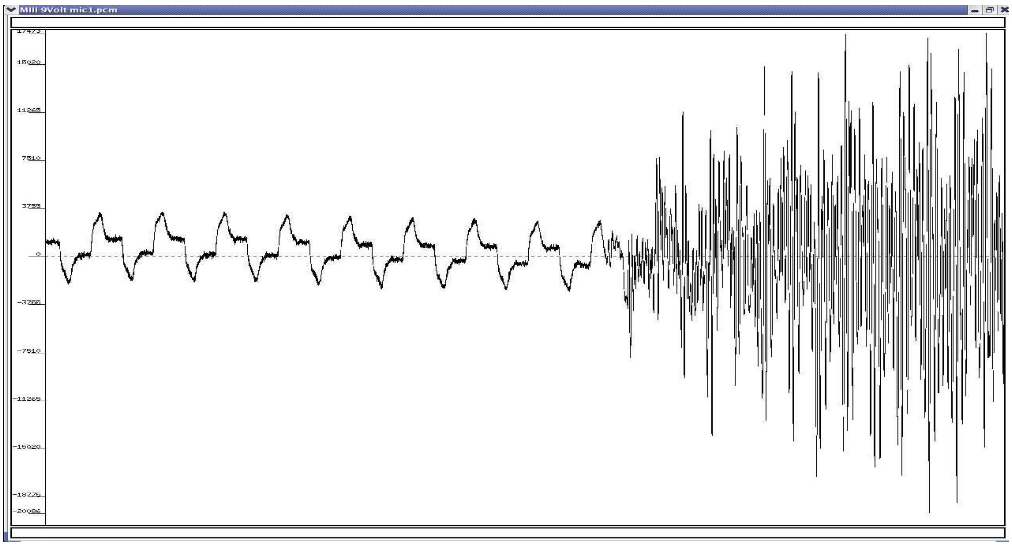

Figure 2: Signal corresponding to a sequence of a silence and a vowel. In the first part of the signal a 50 Hz disturbance is evident.

The 50 Hz disturbance can be seen in Figure 2 (see also attached recordings).

The picture shows clearly the electrical noise (instead of an expected silence signal) in the first part of the signal, while in the second part (representing the beginning of the utterance “A” (phonetic symbol: “a:”)) the speech signal is distorted by 50 Hz; although the latter fact is not so evident in the figure, of course it can be perceived through a listening test. Finally, please note that the periodic signal at 50 Hz has a relatively high magnitude corresponding to about 13 bits out of the 16 bits in the two most significant bytes of each sample.
We had also observed that a Faraday cage helped in reducing this effect; however, the interferences could not be eliminated completely in that way.

## 2.4 Device noise and one-sample delay

The device-noise **represents the major obstacle to the use of the MarkIII for speaker localisation purposes.** It is also subtle to detect, as this problem is neither perceivable in normally reverberant rooms nor evident through waveform or spectral analysis of a single channel.
The device noise problem was evident once eliminated the 50 Hz interference (Section 3.3 reports the way we eliminated it). In other words, the following experiments regard the use of the Mark III array powered by a rechargeable battery and installed in a very quiet insulated room. The room is characterized by less than 30 dBA background noise level (that is very close to the acoustics of an anechoic chamber) and a reverberation time lower than 100 ms. Recordings were done at 44.1 kHz.
As discussed below, the electrical problem can be revealed both at single channel level (perceptually evident through listening tests) and at inter-channel correlation level (through inter-channel coherence measurements) analysis.

### 2.4.1 Single channel analysis

The device noise can be perceptually detected only in recordings taken in a very silent room, because in this condition it can be distinguished from real background noise. Alternatively it can be detected, without the need of an anechoic chamber, by manually detaching the microphones from the boards: the signals acquired from the array is then only pure noise coming from the devices.
We also verified the presence of the same device-noise characteristics on the MarkIII present at UKA. The effect of the device noise can be observed in Figure 3, where two average spectra of 600 ms of silence period are provided. The red line is relative to a single channel of the original MarkIII array, used to record a seminar in UKA on 30/08/2004. The black line is relative to a silence period of the same length we recorded in UKA on 11/11/2004 with the MarkIII modified at ITC-irst. The environmental conditions were approximately the same, but clearly the device noise affects the whole spectrum.
According to the given figures, more than 20 dB noise reduction was obtained at almost all the frequencies. Another very detailed analysis was done by shortcutting each microphone input in order to measure just the board circuitry noise and, also in this case, a noise reduction of about 15-20 dB was observed.

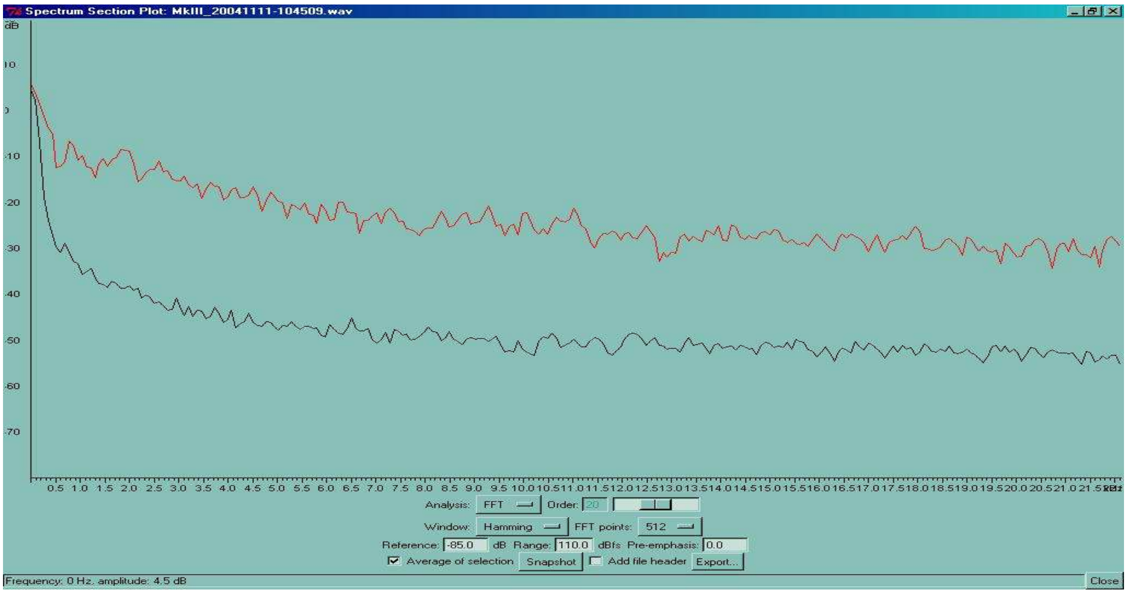


Figure 3: Signals corresponding to average spectra of a 600 ms silence period. The red line hints at the signal quality of the Original-MarkIII, while the black one hints at the signal quality of the Modified-MarkIII. This picture was obtained using WaveSurfer. The signal were sampled at 44.1 kHz sampling frequency and at 24 bit accuracy.

To evidence the problem, we collected other audio data in the above mentioned insulated room.

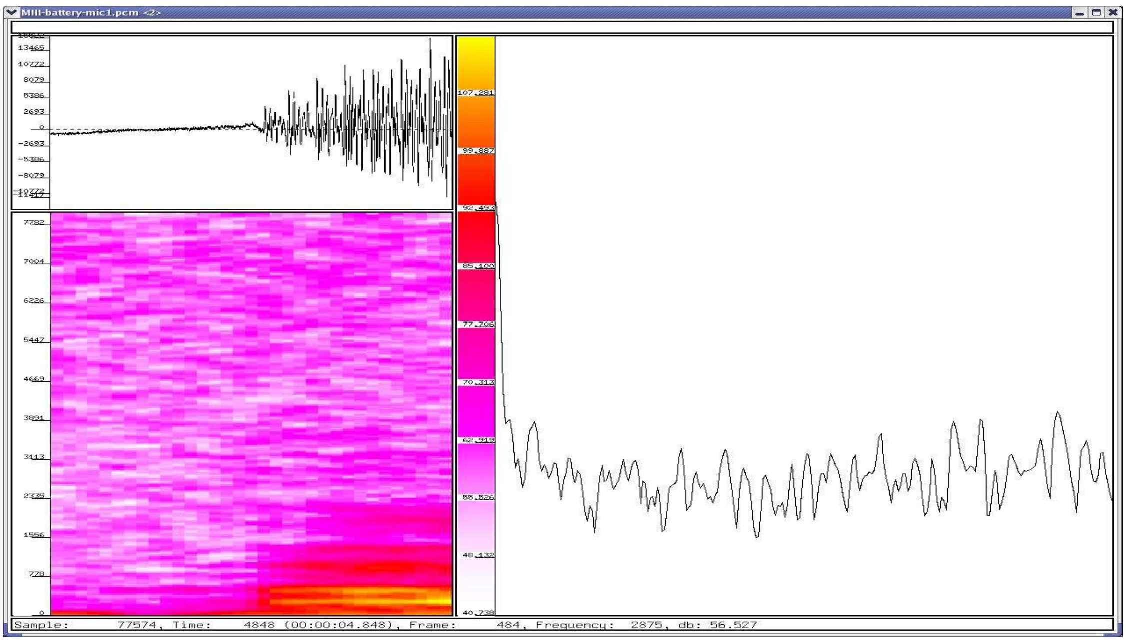


Figure 4: Signal corresponding to a silence followed by a vowel. The lower part of the figure reports the spectrogram. The log power spectrum of the silence segment is given in the right part. This picture was obtained using Sgram software (developed at ITC-irst).

The Figure 4 shows some silence, followed by the beginning of the utterance “A”. As mentioned above, the 50Hz disturbance is not present anymore, the MarkIII being powered by a battery. The spectrum visible on the right part of the same figure refers to that silence period. To better understand the entity of the noise, Figure 5 is related to a zooming (that is a short segment extracted from the first part of the signal of the previous figure).
From Figure 5, one can observe that the noise dynamics (between -300 and +300) involves about 9 bits. It was clear that losing 9 bits out of the first 16 most significant ones was a heavy limitation to the potential of this array.

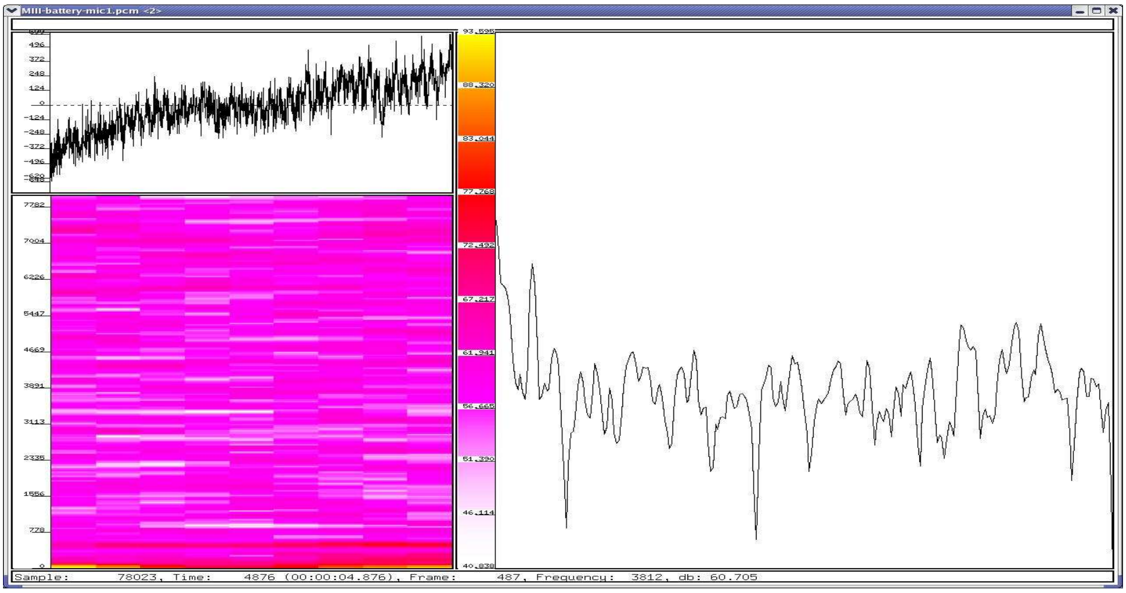


Figure 5: Silence sequence; here, the device noise is more evident both in its dynamics and in its spectral characteristics.

### 2.4.2 Cross-channel analysis

An analysis of the CSP (Cross-Power Spectrum Phase) [3] between pairs of channels put into evidence other problems related to the so called “device noise”.
To interpret the following figures, one has to consider that the mutual delay between microphone pair signals can be associated to a CSP-based Coherence Measure (CM) function $C_{ik}(t,\tau)$ that expresses, for a hypothesized delay $\tau$, the similarity between segments (centered at time instant $t$) extracted from two generic signals $s_i$ and $s_k$. As a result, this function has a prominent peak at delay $\tau$ corresponding to the direction of wavefront arrival. The bi-dimensional CM representation, here used, is derived from CSP-CM in the same way a spectrogram is derived from a FFT.
Coming back to the analysis of MarkIII signals, we observed that the channels of the array are not rigorously synchronous: in particular, sampling instants seem to be multiplexed between odd and even channels. For example, if channels are numbered from 1 to 64, ch2 is in advance of one sample w.r.t. ch1; channel 4 is in advance of one sample

w.r.t. ch3 etc. All odd channels are synchronous, and all even channels are synchronous, but there is one sample of delay of odd channels w.r.t even channels, as shown in the CSP-CM based representation reported in Figure 6, regarding ch1 and ch2 and showing that a peak of CSP is constant at one sample delay.

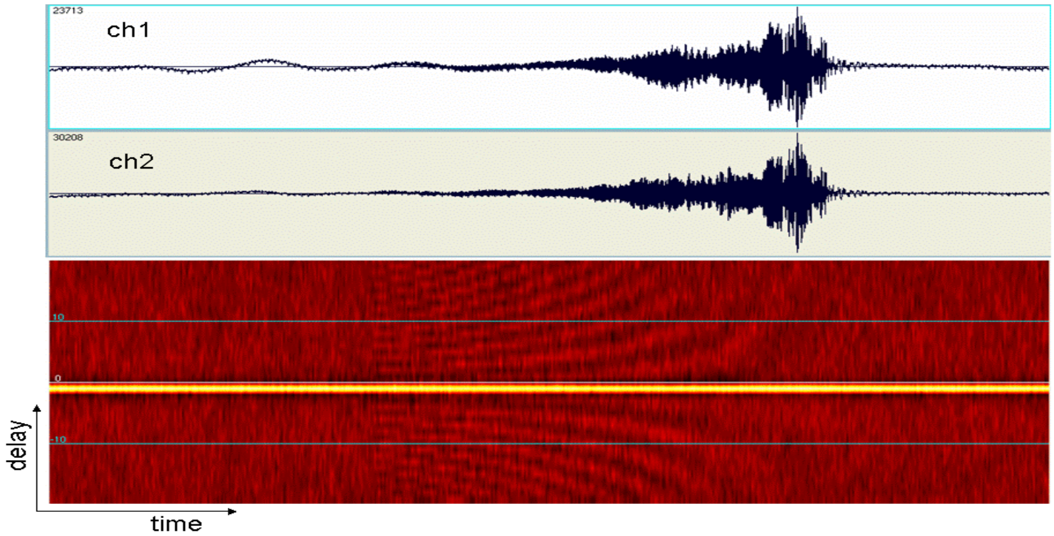


Figure 6: Signals extracted from Channel 1 and Channel 2. CSP bi-dimensional representation is reported in the lower part of the figure with a “heat” palette black-red-yellow- white from low values to high values.

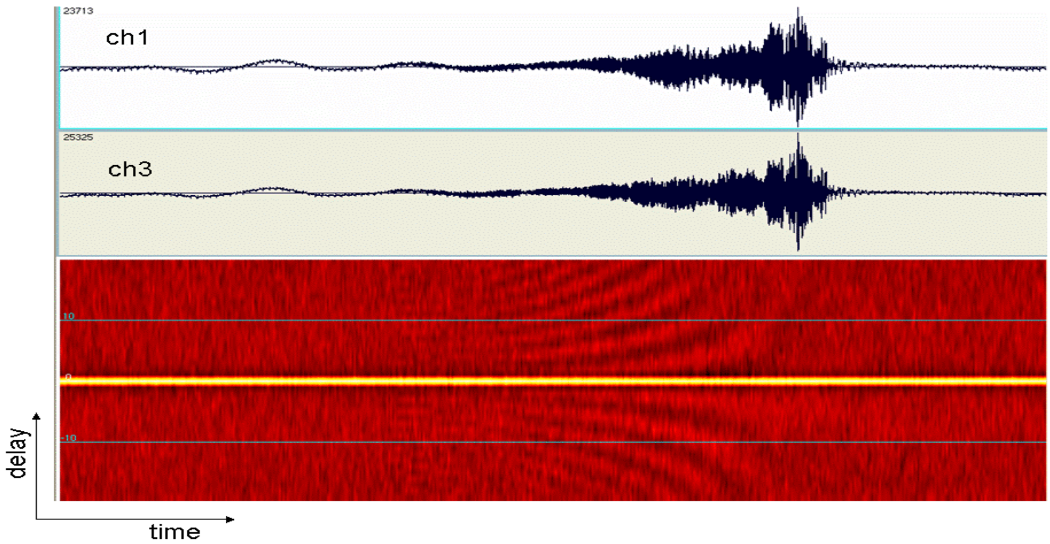


Figure 7: Signals extracted from Channel 1 and Channel 3. The peak of the CSP function reported in the lower part of the figure is now centered on 0 samples.

On the other hand, the same CSP analysis on ch1 and ch3 shows no offset, as depicted in Figure 7.
Moreover, as introduced in paragraph 2.4.1 for what concerned a single channel analysis, a strong noise component (at least 10 bits out of the 16 most significant bits) can be observed in all the channels, which is neither acoustic noise nor transduction noise of the

microphones. It dominates over acoustic background noise of a relatively quiet environment. This noise has a “common mode” within the 8 channels of each array micro-board. Different modules (e.g ch1-8 and ch9-16) have different and uncorrelated noise components. This is evident again on the basis of a CSP analysis. Figure 8 shows the noise coherence between ch1 and ch8: in this case, a strong coherence is evident between the (mainly electrical) noise sequences.
On the other hand, the same analysis repeated on channels ch1 and ch9, which are on two different microboards and therefore have no common mode noise, demonstrates the absence of any coherence at any particular delay, as shown in Figure 9.

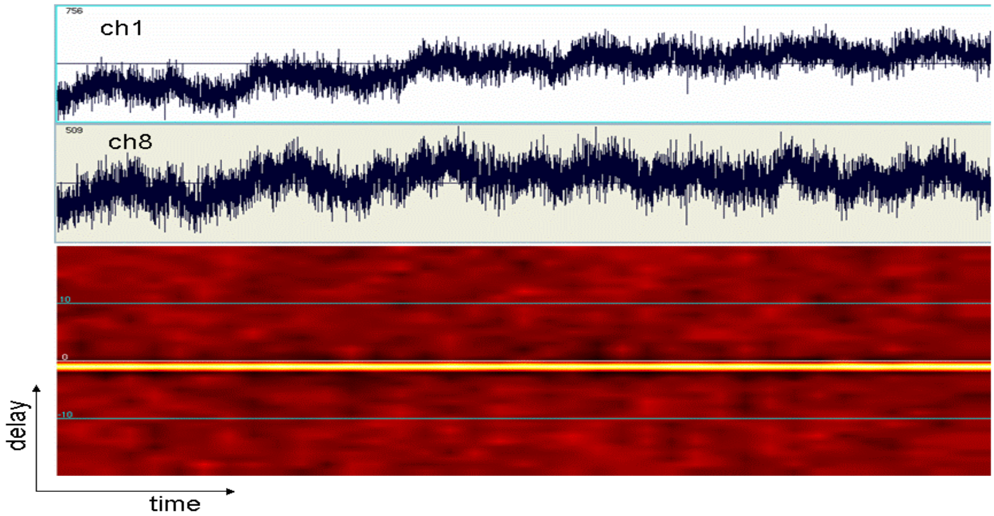


Figure 8: Signals extracted from Channel 1 and Channel 8. The peak of the CSP function reported in the lower part of the figure shows a strong coherence between the device noise sequences.

So, only a maximum of 8 microphones (one per micro-board) in an array of 64 microphones are entirely uncorrelated, thus reducing dramatically the usefulness of 64 microphones and related bandwidth.
To this regard, it is worth noting that **any localization technique as well as many beamforming algorithms would find an artificial coherence at zero (or one), leading to the hypothesis of a source in front of the array (at an “infinite” distance) any time there is not a more dominant speech source (e.g. a speaker having a relatively loud voice).** In practice, while a speech source is active, the effort to locate the speaker would be vanished by a bias introduced by the device noise across the channels.

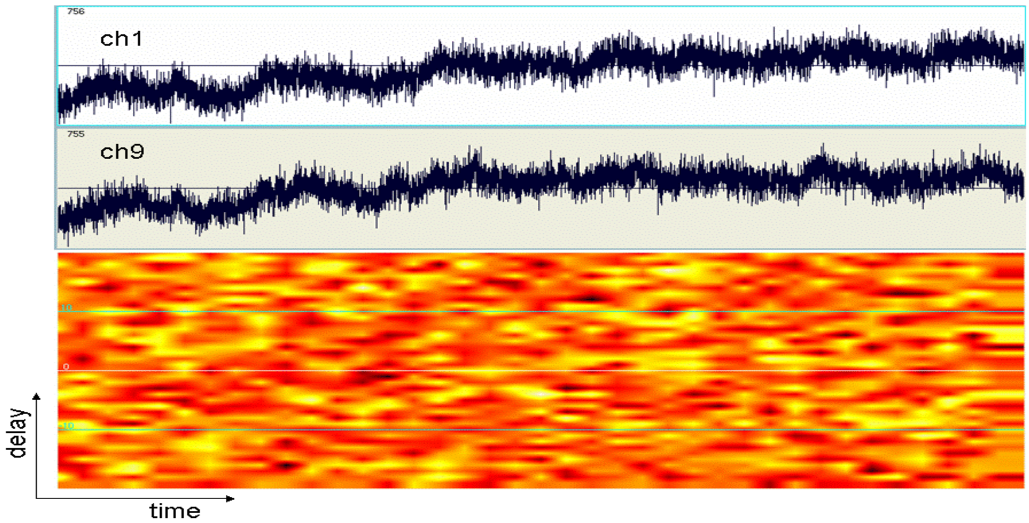

Figure 9: Signals extracted from Channel 1 and Channel 9. The peak of the CSP function reported in the lower part of the figure shows that there is no coherence between the two device noise sequences.

## 2.5 8 and 16 kHz common ground noise

We observed a further problem by analyzing the spectrogram of some utterances. This problem became evident once both the 50 Hz and the device noise problems were solved. Two disturbances at about 8 and 16 kHz appeared in the spectrogram, as can be seen in Figure 10: two relatively strong stripes appear in red in the spectrogram on the left part of the picture, which correspond to the two peaks evident in the right part.

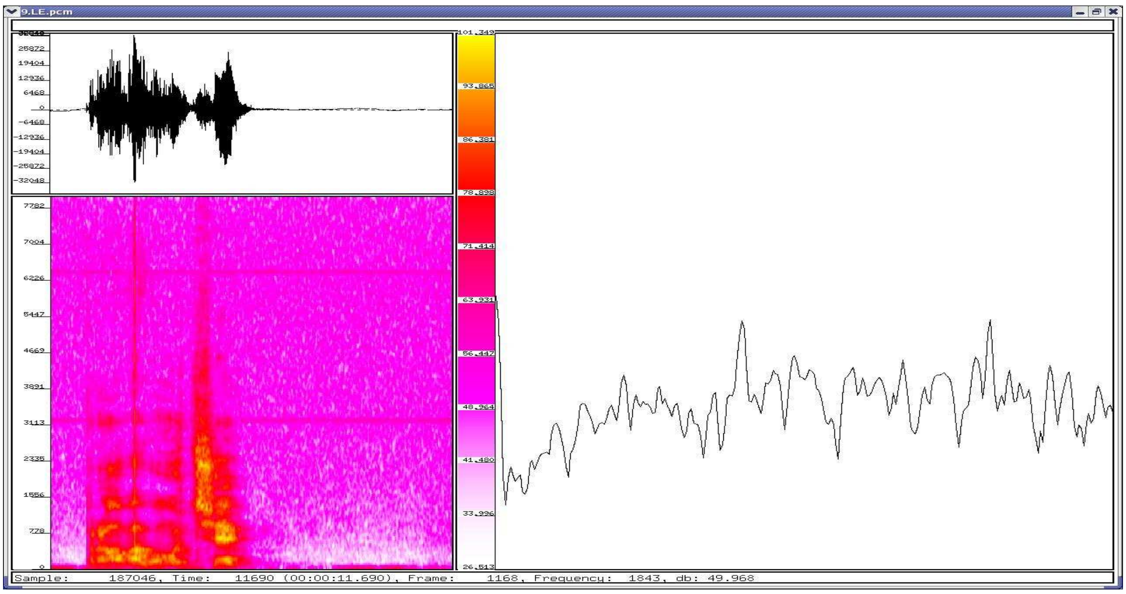

Figure 10: The 8 and 16 kHz disturbance peaks are evident in the right part of the picture, where the spectrum of a silence instant is taken after the utterance depicted in the left part. Notice the absence of the device noise (that was eliminated as discussed in Section 3.4).

Though the disturbance is present at frequencies not closely related to the speech signal, we verified it does not come from the environment and it was then worth to investigate, as it represents another common mode noise component across different channels.

### 2.6 Anomalous behavior at 22 kHz

While measuring Impulse Response (IR) in our pseudo-anechoic chamber we noticed that the same algorithm used to calculate the room IR provided accurate IRs with chirp signals (see in [4] the use of chirp signals for impulse response estimation and sensor calibration purposes) recorded at 44.1 kHz, while it gave IRs of unusual characteristics when acquiring chirps at 22.05 kHz: in fact, they showed increasing oscillations for about 50 samples before the main peak occurs (observable by comparing Figures 11 and 12).

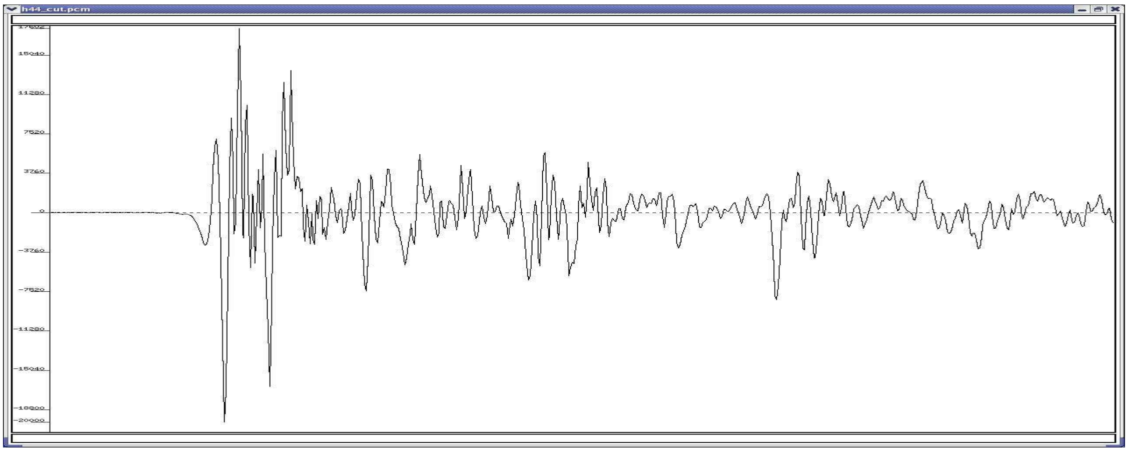

Figure 11: Impulse response of the IRST pseudo-anechoic chamber at 44100 Hz. The figure depicts the first 15 ms of the IR. Chirp signals to calculate IRs were acquired prior to our intervention on the array. The main peaks are evident.

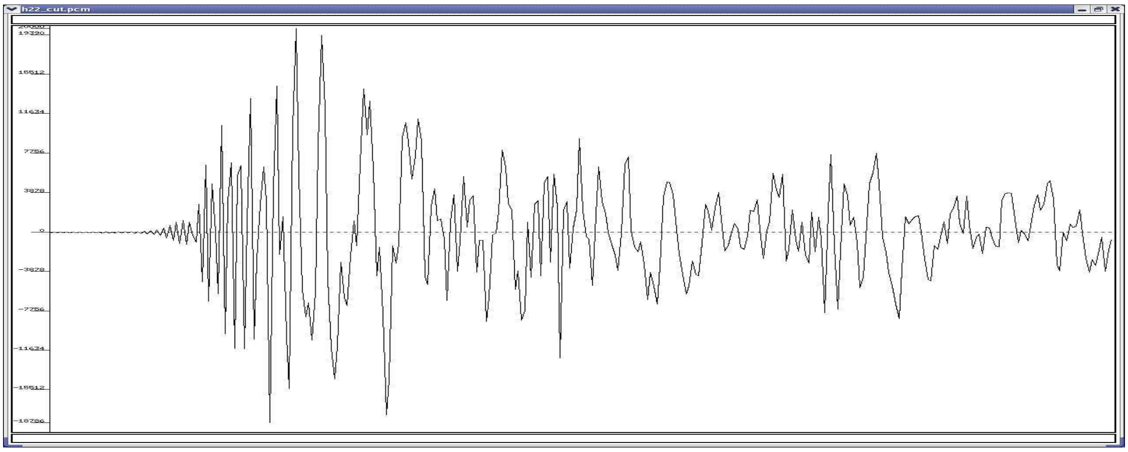

Figure 12: Impulse response of the IRST pseudo-anechoic chamber at 22050 Hz. The figure depicts the first 15 ms of the IR. Chirp signals to calculate IRs were acquired prior to our intervention on the array. The main peaks are not evident at all and an oscillation with an exp(x)sin(x) behavior appears at the very beginning

The problem is still unsolved, and by now it shows that recordings at 22.05 kHz might not be reliable.

### 2.7 Potential micro-boards breakdown

The designer of the MarkIII array reported the breakdown of some microboards after many hours of acquisition. This leads to the conclusion that the MarkIII stops working after some time. The changes proposed in Section 3 solve also the latter problem, as addressed in Section 3.6.

## 3 Modified MarkIII: hardware modifications for the first improved version

### 3.1 Early saturation

We initially thought the problem was unsolvable, since no trimmer or other regulations of the input level are available. We then decided to physically bypass the first amplification stage as described in the following.

We substituted the two capacitors, placed at the very beginning and at the very end of the amplification stage, with two polyester 0.47 uF capacitors, which generate much less noise. We then chose to limit the amplifier gain: we thought the first stage gain was 100 and the second one was 6.8 (as reported in [1]), so we bypassed the first stage via a 0.47uF capacitor, keeping the second stage polarization to the phantom GND with a 100KOhm resistor.

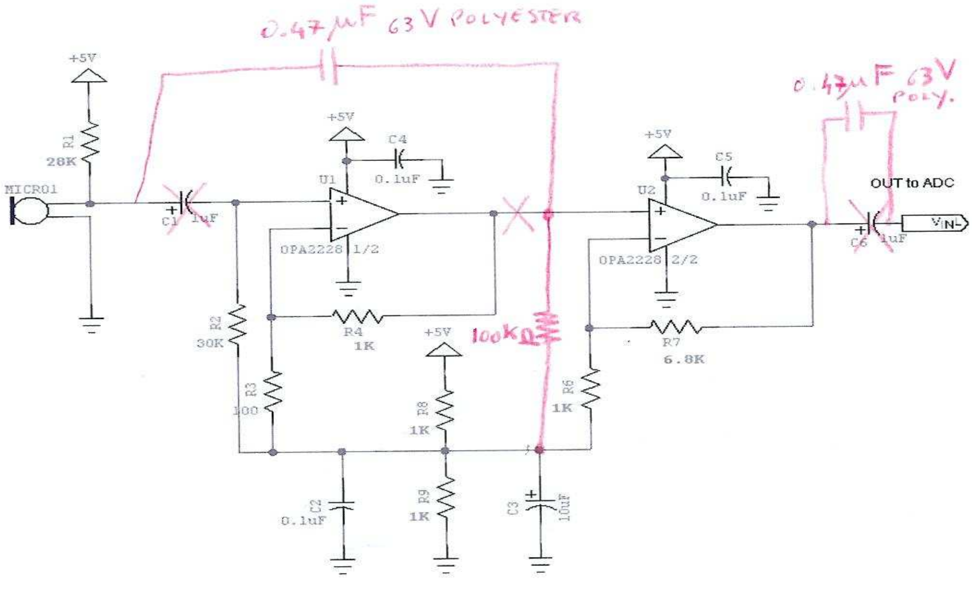


Figure 13: This figure shows the modification on the amplification stage relative to one microphone.

We then realized the first stage gain is actually 10. The original gain of 68 was reduced to 6.8, as depicted in Figure 13, but one could also keep the first stage rather than the second one

However, we will eventually propose a new solution in order to better exploit the dynamic range, as a 6.8 gain is sometimes too low if the speaker has got a weak voice: we will choose the value which statistically prevents as much as possible the clipping effect and we will consequently change the ratio between the resistor values in the reaction network around the amplifiers. We expect that a gain around 15-20 could be an optimal final choice.

### 3.2 Packet loss and other inconsistencies

The packet loss problem can be solved at the client side via software: the Smartflow client, developed at NIST and improved at UKA, allowed us to record a 40 minutes seminar losing only one frame. This is because the improved version makes use of two threads (the first to acquire form the array, the second to write on disk) and one ring buffer.

For what concerns the inconsistencies in the beginning of the recorded signals, as the first 4350 samples (at most) can include either a sequence of zero samples or a short sequence of random samples or a spike at the very beginning, we suggest to simply remove them from every channel.

### 3.3 50 Hz disturbance

We realized that the disturbance was due to the power supply provided with the array and we solved this problem by substituting the 9V power adaptor, provided with the array, with a Pb rechargeable battery. When we also solved the device noise problem we adopted a different solution and we designed a power supply block which fed the analog part of the array with batteries and the digital part with a transformer, as specified in the next section.

Furthermore, even with the best power supply available, still a light 50Hz disturbance persists: it is much lower than the one coming from the power supply and it is totally due to environmental electromagnetic fields. By consequence it can be definitely eliminated by surrounding the MarkIII with a Faraday cage.

### 3.4 Device noise and one-sample delay

The device noise was, as we suspected from the beginning, caused from the tension regulator LM2940 (see technical documentation of Mark III in [1]). There is one such a regulator for each of the 8 microboards. This tension regulator provides the operation voltage to 8 Panasonic microphones, to 4 A/D converters and to 8 OPAMPs. As mentioned, the device noise has a common mode within the 8 channels of each array microboard: by deduction, either this noise comes from the motherboard, or it has to be generated by a component which serves ALL the microphones or OPAMPs or A/D converters of the same micro-board, so it cannot be a specific OPAMP, a specific A/D or a passive component around them. We excluded the first case, as from the motherboard to the micro-boards only 3 clocks and DATAS are exchanged, together with power supply. Clocks are in common for 4 microboards out of 8, but still the noise was different between them. Data are in digital format and disturbances do not cause discontinuities in the signal.

In order to keep the original device layout, the problem has been solved by physically removing such regulators and feeding the analogue part of every board directly with a circuit of battery designed ad hoc, while the digital part remains fed with a new transformer stabilized and filtered ad hoc. This implied an analysis of power consumptions prior to any decision about the components to buy: this analysis is provided in Section 3.4.2.

Finally, we can observe that for each couple of mics the same A/D converter is used: we suspect that the A/D circuit is responsible of the one-sample delay. In practice, there is no compensation to the fact that the A/D serves, in turn, the odd and the pair microphone, respectively, unless this is done via post-processing the channels or by re-designing the micro-boards.

### 3.4.1 Device-noise: history of interventions

The device-noise is mainly caused by the tension regulator LM2940: let us analyze the circuitry around it.

The surface-mounted polarized capacitors (generally used because they are small), placed after the regulator in the original design, do not seem to be suitable for audio application: they have an inner leakage current which creates the necessary oxide between the armors, thus generating a disturbance. We substituted the polarized capacitors with polyester capacitors, which are bigger but generate much less noise, as depicted by Figure 14.

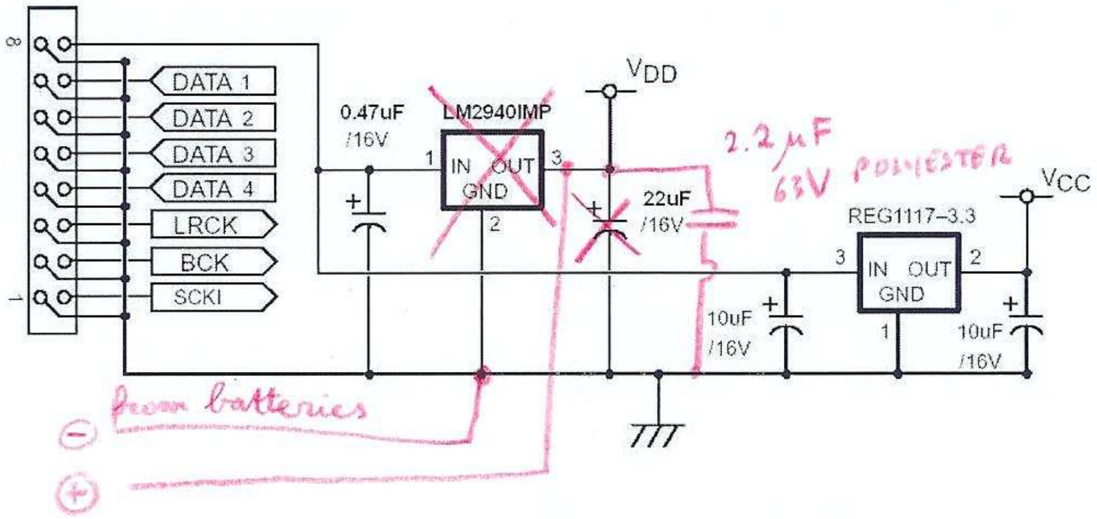


Figure 14: Schematic of the modifications in the motherboard connection stage.

In synthesis, part of the device-noise is caused by the LM2940, which is not eliminated by the polarized capacitor, and part by the polarized capacitors themselves.

A first solution we envisaged was putting an LC cell after each regulator in order to have a DC free from disturbances, but the results are much better by bypassing the regulators and providing power supply to the digital part of the microboards with batteries. The power supply box has been designed by considering the following specifications:

- A/D PCM1802 power supply requirements: (from the datasheet):
  - Vcc: nominal 5V, minimum= 4.5V, typical= 5V, maximum=5.5V
  - Vdd: nominal 3.3V, minimum= 2.7V, typical= 3.3V, maximum=3.6V
- OPAMP OPA228 power supply requirements: (from the datasheet):
  - Vs= minimum +-5V, maximum +-15
- Panasonic microphones require from 5V to 12V. The voltage is limited by a high value resistor.

We substituted the 5V regulator with a non-rechargeable battery: we put 3x4.5V Zn-C batteries in parallel to feed 3 microboards. The artificial correlation effect, which was visible via CSP [3], disappears (see Figure 15). To make this improvement practical, we then switched to a combination of 4x1.2V Ni-Cd batteries in series: they give the best performances. We later realized that feeding more than one microboard with the same group of batteries, or equivalently with the original 9V transformer, creates the 8 and 16 kHz disturbance (see Section 3.5).

We concluded that the best solution was to feed the analogue part of each microboard with 4 x1.2V, 5Ah batteries, so to guarantee the galvanic de-coupling of each power supply source.

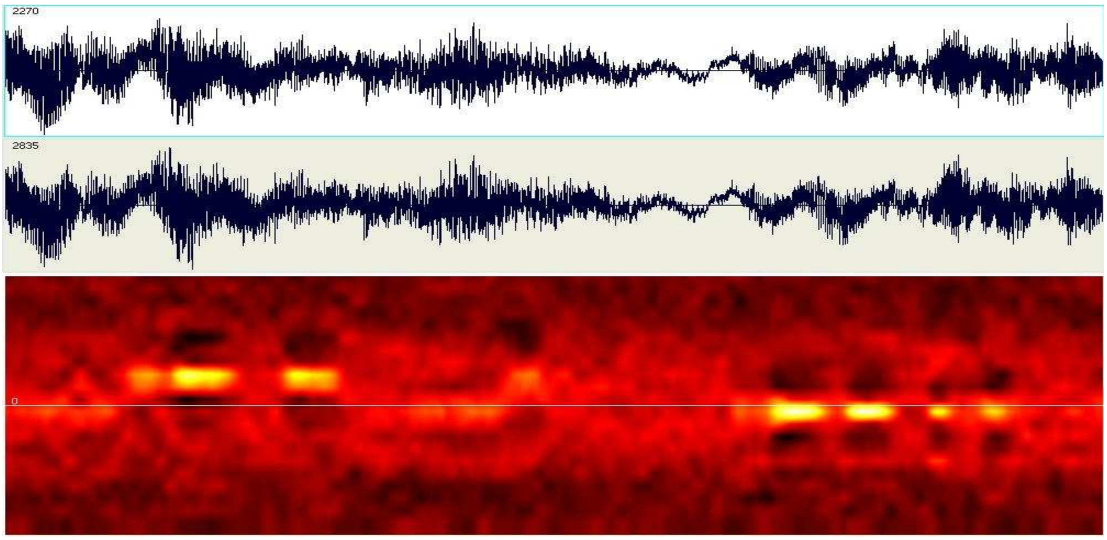


Figure 15: Signals extracted from Channel 1 and Channel 8. The peak of the CSP function reported in the lower part of the figure shows a strong coherence only when either one speaker (in the left side) or another speaker (in the right side) are talking.

### 3.4.2 Power Consumption

When we bought the MarkIII we roughly measured a 1A consumption. The array actually consumes more in acquisition state. A more precise measure is needed after the choice of

including batteries in the power supply stage. We observed that the motherboard absorbs 200 mA, while each microboard absorbs 100mA for the digital part. We have 0.1A x 8 + 0.2A = 1A required. The analogue part consumes exactly 180mA per microboard, thus 0.18A x 8 = 1.44A required to the Ni-Cd batteries. The total consumption is then 1A+ 1.44A = 2.4A (measured with a tester while acquiring).
The power supplier we designed and realized is able to provide 1.5A at 9V: for the sake of practice we used the same transformer both to recharge the Ni-Cd batteries when not acquiring, and to feed the digital part when acquiring. The 32 batteries are set in 8 groups, connected in series: they provide 5A/h each, i.e. 0.2A for 18 hours with an efficiency of 70%. The recharge lasts 15 hours. So, in the worst case, we theoretically should make the MarkIII run for 20 hours without interruption.

### 3.5 8 and 16 kHz common ground noise

This problem is well visible in any very silent environment and prevents any clean data collection: we discovered it was due to the coupling between the digital and the analog ground. This coupling was made around the PCM1802: the device is originally provided with two separate pins for the two grounds. In the original project of the MarkIII the two pins were connected via a short circuit. This makes the analog ground, which the audio signal relies upon, coincident with the digital ground, which collects the noise coming from the various integrated devices, such as the A/D converter and the two tension regulators.
We tried to solve this problem by inserting a LC cell after the alimentation or, alternatively, by inserting a resistor in series, without any success: with the LC the strong 8-16 kHz disturbance was still present, while with resistor dropped down the power supply voltage and the MarkIII was not able to acquire. The final solution consists in avoiding the common ground by feeding each micro-board separately with an independent group of batteries.

### 3.6 Potential micro-boards breakdown

This problem is caused by the polarized capacitor immediately adjacent to the microphone: the network of the amplification stage is built so that when the MarkIII is on there is always a tension around the capacitor. This tension after about 10 hours breaks the capacitor, thus preventing the array from acquiring anything. We did not experience the problem because we immediately substituted all the polarized capacitors with polyester capacitors.

## 4 Possible future improvements

As the modifications here reported lead to a new version of the array based on an external box for power supply, it was observed that its reproduction implies problems of cost and portability.
Hence, a different set of modifications, not requiring the external box, is under study, to avoid using a distinct rechargeable battery for each part of the circuitry. Although it is

worth noting that the foreseen power supply system may imply more noise than what offered by the here presented solution, we are looking for a reasonable compromise between the two opposite needs of portability and of immunity to electrical interferences.

Beside that, other possible interventions will regard a change of amplification gain and the introduction of a post-processing for dynamics control.

## 5 Conclusions

The present draft illustrated the problems encountered in using the MarkIII array as well as the suggested modifications to derive a new device characterized by a much higher quality in the acquired signals. The main principle for this partial re-design of the device was related to the objectives and the needs arisen for the research activities conducted inside the CHIL E.C. project. However, the proposed modifications represent a necessary step for anyone who will use this device for any activity based on the most traditional microphone array processing techniques.

To summarize, the main changes on the device were caused by the following choices in the original design:

- the choice of the LM2940 tension regulators, which cause the common-mode noise on the 8 microphones afterwards.
- the choice of polarized capacitors in some points of the micro-boards, which increase, instead of decrease, the noise generated by the LM2940 and causes a potential breakdown of the micro-boards.
- the choice of the power supply connector on the motherboard, which causes false contacts.
- the choice of an amplification gain equal to 680 (recently measured as 68), which causes early saturation. This anyway depends on the application and on the environment.
- the short-circuit between analogue and digital ground, which causes high frequency disturbances.

Although the MarkIII device acquires both at 22.05 kHz and ar 44.1 kHz, the proposed modifications here exposed will allow to acquire at 44.1 kHz with a much higher SNR but leave the use at 22.05 kHz open to further investigation. In fact, the impulse responses measured with a chirp signal acquired when the MarkIII is working at 22.05 kHz are still anomalous.

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## Appendix A: List of components necessary to modify the MarkIII

| RS code | Component | Quantity |
|---|---|---|
| 467-8461 | Capacitor polyester 0.47microfarad | 128 (i.e. 26 boxes) |
| 467-8483 | Capacitor polyester 2.2 microfarad | 8 (i.e. 3 boxes) |
| 148-972 | Resistor 100kOhm | 64 (i.e. 13 boxes) |
| 231-8375 | Transformer 100VA, prim. 115V, sec. 15 V | 1 |
| 159-900 | Resistor 10 ohm | 8 |
| 264-901 | Bridge rectifier | 3 |
| 432-6867 | Connector | 1 |
| 377-7781 | Battery NiCd | 32 |
| 326-6736 | DPCO Relays (MY2 Series) | 4 |
| 196-6147 | Relay sockets | 4 |
| 359-5955 | Fuseholder | 7 |
| 432-6845 | Connector | 1 |
| 127-486 | Compact Electolytic Capacitor 22000 uF | 1 |
| 228-6824 | Electrolytic Capacitor 2200 uF | 1 |
| 211-5047 | Radial,Multi Layer Ceramic Capacitors | 1 |
| 461-9131 | PCM1802DB | 4 |
| 410-8827 | Connector for resolver | 1 |
| 410-8990 | Connector for resolver | 1 |
| 453-678 | Tini Q-G Min Audio Connectors 4 poles | 2 |
| 221-4885 | Soder-Wick Desoldering Braid 0.9 mm | 1 |
| 221-4891 | Soder-Wick Desoldering Braid 1.5 mm | 1 |
| 229-4263 | Multicore Crystal - No Clean 250g | 1 |
| 399-596 | Black and Pink Sleeves H20 | 1 bag |
| 399-603 | Black and Pink Sleeves H30 | 1 bag |
| 270-4215 | Cable 16x 0.25 | 3 meters |
| 178-490 | Commercial White Nylon Cable Ties | 1 bag(~100 units) |
| 293-7645 | Audio cable red and black | 1 |
| 316-838 | Toggle Switches, Std, 15A DPST | 2 |
| 177-5288 | Linear Voltage Regulator 9V | 1 |
| 648-438 | Linear Voltage Regulator 12V | 1 |
| 348-5397 | Standard Rectifiers 1A to 6A | 4 |
| 738-402 | Pole Terminal, 63A | 1 |
| 419-779 | Min 5x20mm Anti-Surge Fuses 2A | 1 |
| 237-0519 | Panel Mount LED Holder | 2 |
| 590-446 | 5mm Std & Intensity Matched LED | 1 |
| 222-070 | Two Tone Grey Housing | 1 |

**Appendix B: Figures about signals and interventions**

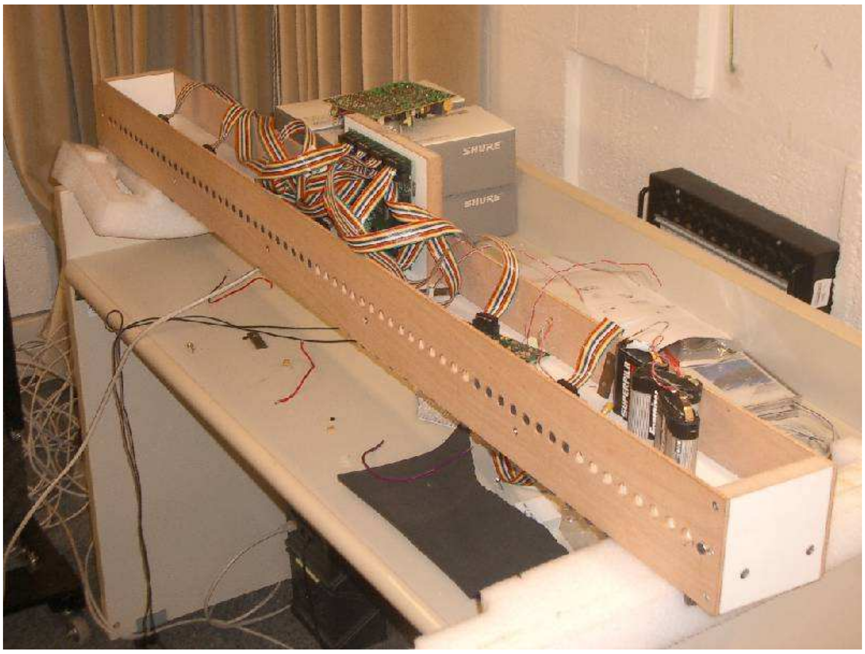

**Figure B-1: The MarkIII during our interventions: we built a wooden cage around it and we verified it does not affect results.**

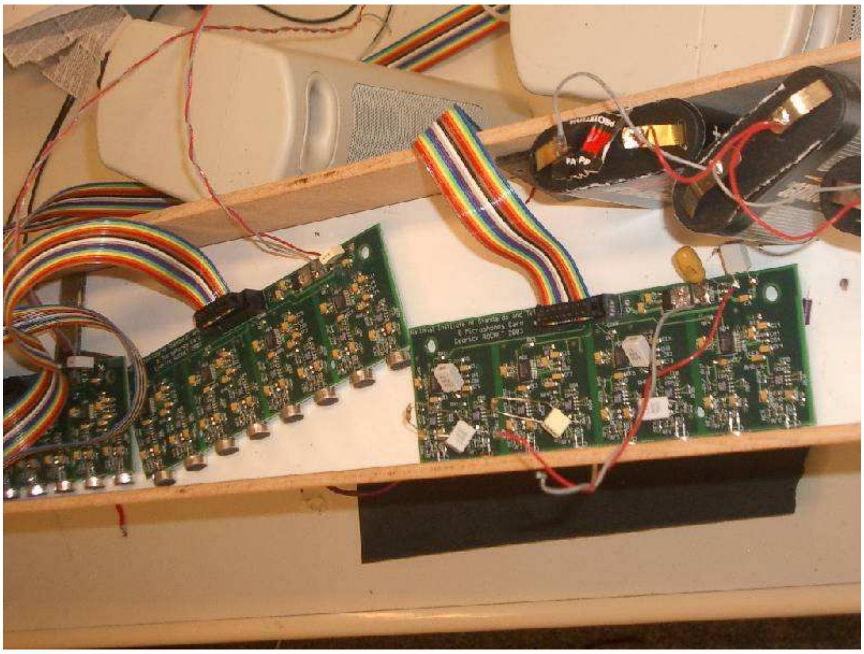

**Figure B-2: One of the micro-board is being modified: some polyester capacitors are placed instead of the polarized SMD ones in the amplification stage and around the tension regulator LM2940, which was removed. Some non-rechargeable batteries provided power supply at that time for the analog circuitry.**

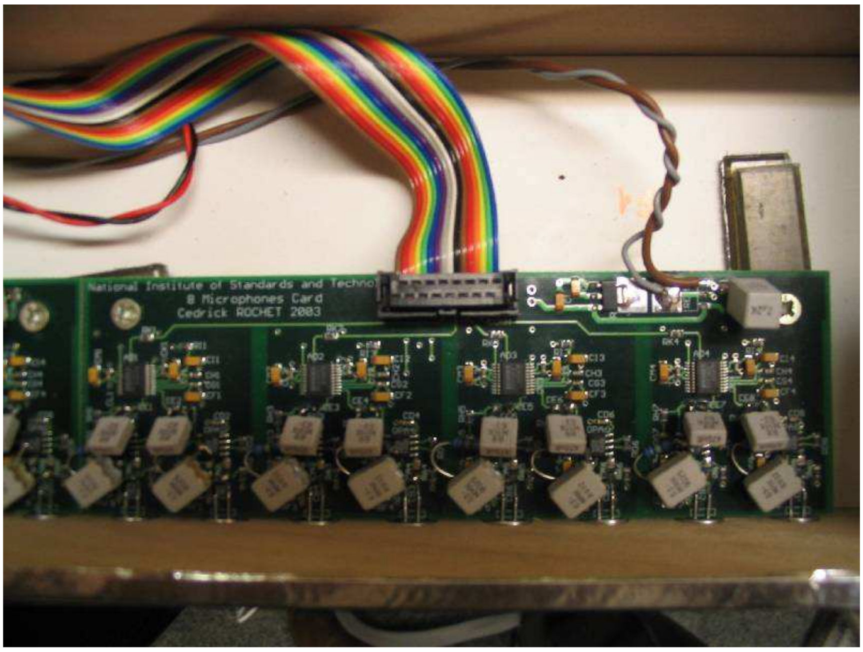


**Figure B-3: The micro-boards after our intervention: for each microphone two capacitors and one resistor were used, plus one capacitor for the alimentation which comes from the red and grey twisted cables. Notice that the use of the latter capacitor prevents the potential micro-board breakdown problem.**

**Figure B-4: the IRST-MarkIII after the intervention: an additional Faraday cage was added to isolate the last 50Hz noise residual.**

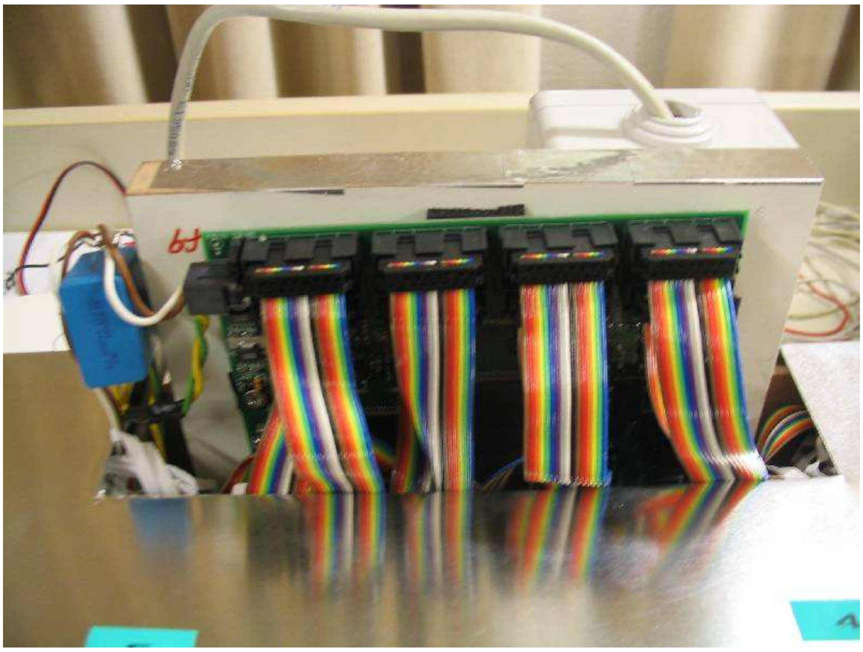

**Figure B-5: The motherboard of the IRST-MarkIII. It is by now only half electrically separated from the micro-boards. Our future interventions will isolate completely the motherboard.**

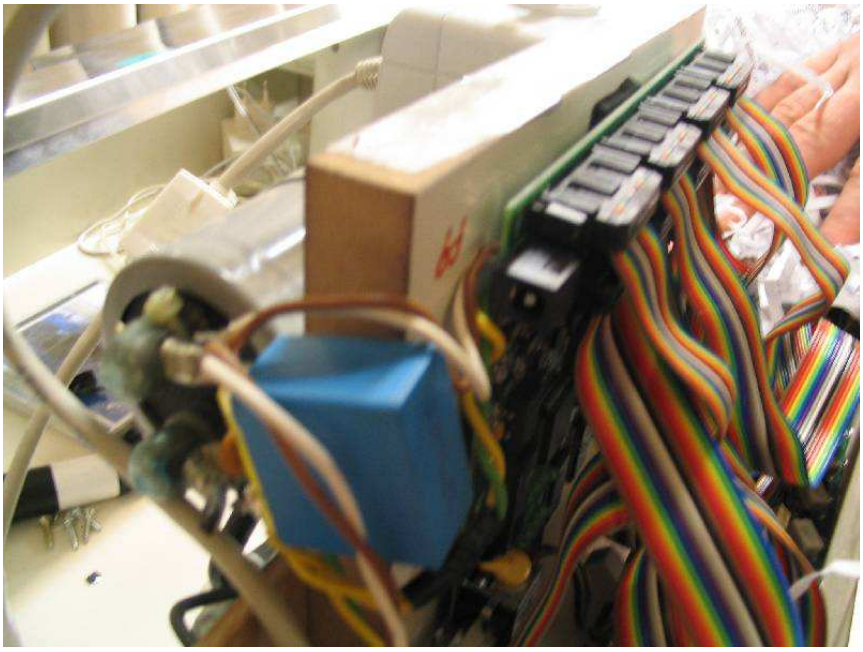

**Figure B-6: Zoom of the motherboard. Note that the power supply (brown and white cables) skips the original black connector to avoid false contacts. The additional capacitors (gray and blue) are also visible: their purpose is to low-pass the power supply current so that it comes to the motherboard almost free of disturbances.**

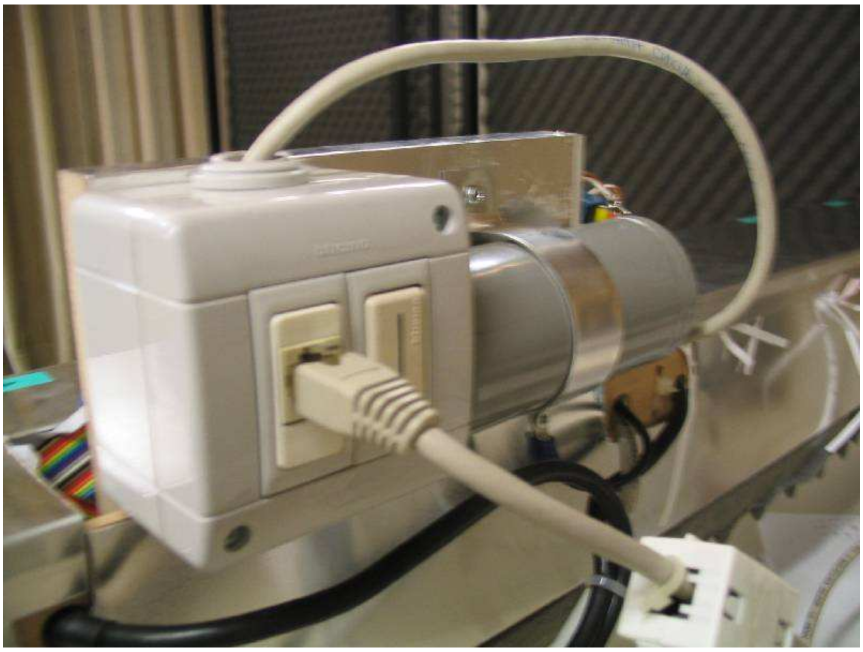

**Figure B-7: The motherboard from behind: an Ethernet socket has been added for practical purposes only. The big 15 mF condenser is also visible.**

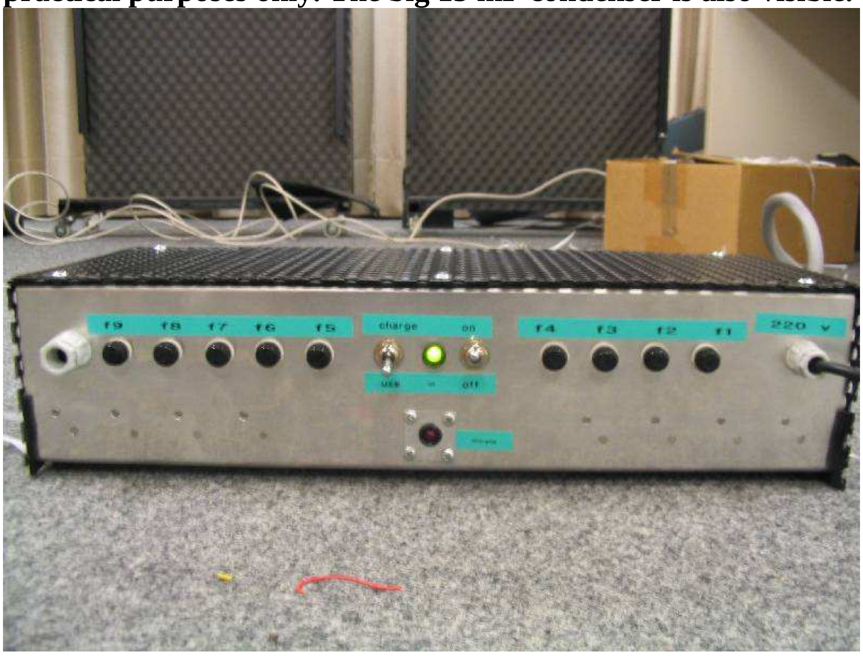


**Figure B-8: The front of the power supply box entirely designed and built at ITC-irst: the 220V power supply input, the fuse protecting the micro-board alimentation, the green led (acquisition state) and the red led (recharging state) as well as the two toggle switches are shown.**

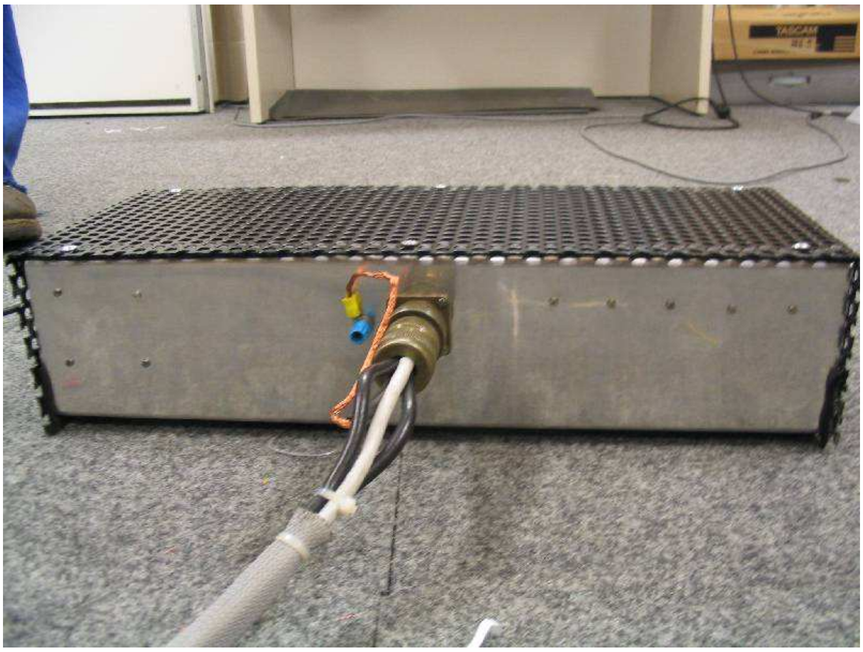

**Figure B-9: Back of the power supply box: the ground is connected separately. The box is surrounded by a metal cage.**

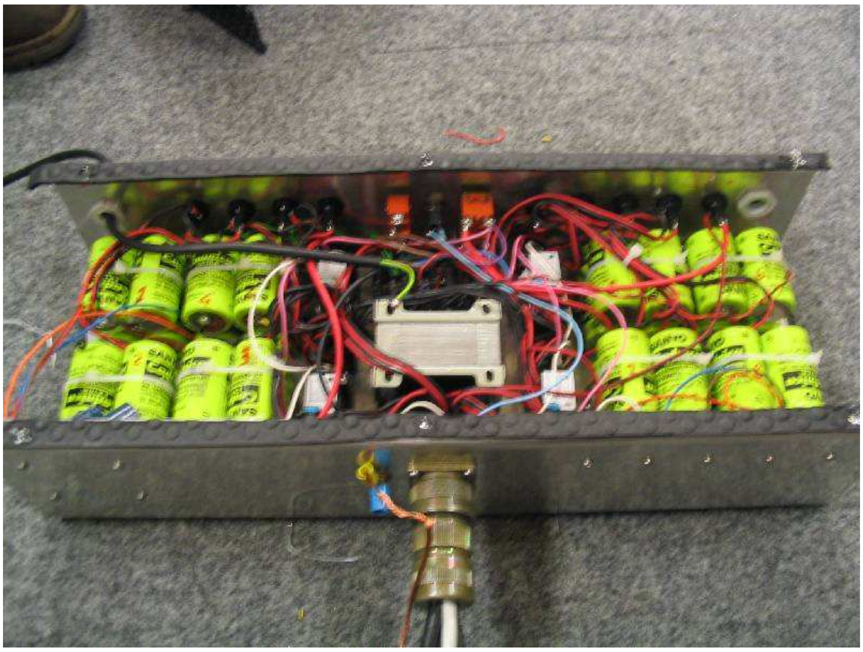

**Figure B-10: Inside of the power supply box: from the 8 groups of 4 batteries the power supply passes through the red and violet cables, placed on purpose in those positions. The transformer which provides power supply for the digital part (in acquisition state) and recharges the batteries (in recharging state) appear in the center of the box.**

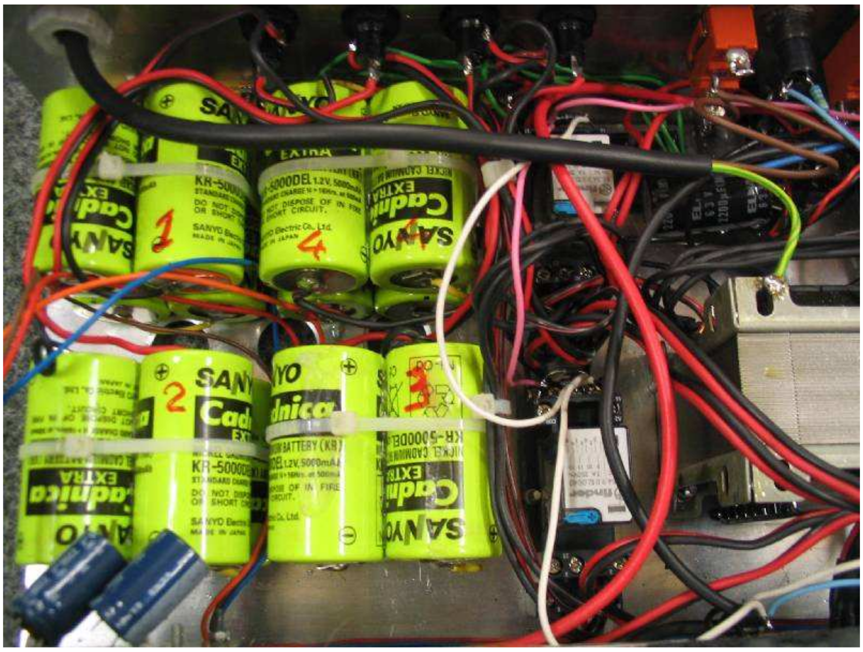


**Figure B-11: Particular of the power supply box: battery groups 1 to 4 are shown together with tension regulators and 2 relays.**