

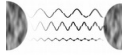
TECHNICAL REPORT

Fredrik Stenmark

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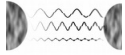
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1 INTRODUCTION

The purpose of this document is to provide a detailed specification of the system, including both functionality and implementation details. The system demonstrates some of the powers of massive MIMO beamforming, namely that signal energy can be focused to a specific point of interest and at the same time be suppressed in areas around this point.

1.1 MIMO

MIMO (Multiple Input Multiple Output) technology has gained an increasing interest in wireless communication due to its capability to increase the throughput of a communication system without an increased bandwidth or increased transmit power. The technology is based on the use of multiple antennas, both in the transmitter and the receiver. By using equal transmit power on all antennas, a higher spectral efficiency is achieved (bits per second per Hertz of bandwidth) for the same transmitting power due to the resulting array gain of the antennas.

1.2 Beamforming

The MIMO technology allows to direct signal power in a certain direction by altering the phase of the transmitted signal on each of the antennas in order to create constructive interference in a certain point or direction. This enables a transmitter, a base station for example, to direct the communication to a certain receiver without interfering with other receivers.

1.3 The Project

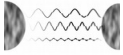
In the course TSKS05 – CDIO Communication Systems – the students were supposed to construct a system that would demonstrate the powers, described in this introduction, of massive MIMO beamforming. By using a set of loudspeakers, that can also be used as microphones, the students were to build a system that was able to measure sound transmitted from a source and from that sound extract characteristics of the channel between the transmitter and receiver. With this information the system can produce a sound through the loudspeakers, and the sound will be constructively added at the point, where the measured sound originated from. The original project idea was to place a set of four wine glasses next to each other, ringing one of them and letting the system crack this glass, leaving the remaining glasses unaffected. This would be a spectacular way of showing the powers of massive MIMO, which the product aims at demonstrating, since it becomes rather obvious that signal power is focused into one point and suppressed essentially everywhere else. However, since the product is limited to only eight pairs of low powered loudspeakers, the product is not able to produce enough signal power to crack a wine glass. Therefore the more controlled and secure product setup, described in Chapter 2, is used.

1.4 Aim and Goal

See Section 2.1 in the document Project Plan [1].

1.5 Definitions

Word	Definition
MIMO	Multiple Input Multiple Output
A/D	Analog to Digital
D/A	Digital to Analog
L/M	Loudspeaker and microphone unit
L/M-pair	A pair of loudspeaker and microphone units
OS	Operating System



Word	Definition
Subsystem	A part of the whole system
DB-9	D-sub connector with 9 pins
DB-37	D-sub connector with 37 pins
Loudspeaker mode	The mode of an L/M-unit where it functions as a loudspeaker and can transmit sound
Microphone mode	The mode of an L/M-unit where it functions as a microphone and can record sound

2 OVERVIEW OF THE SYSTEM

The system consists of two subsystems, software and hardware. The user interacts with the software, which communicates with the hardware, in order to demonstrate that signal energy can be focused into a very precise point and at the same time be suppressed essentially everywhere else around this point, using massive audio beamforming. The product setup can be seen in Fig. 1. The software subsystem is hosted on the computer and the software design is specified in Chapter 3. The hardware subsystem includes everything in the product, that is not included in the software subsystem, and is further described in Chapter 4.

As seen in Fig. 1, a total of eight L/M-pairs are used in the product. The L/M-pairs are connected to a distribution box, whose task is to provide the correct connections. The distribution box is in turn connected to the A/D and D/A converters, mounted in the computer. When the system operates, seven L/M-pairs is first used to receive sound data that is sent from the L/M in the point of focus. That is, from one of the L/Ms in the eighth L/M-pair. Then the received sound data is analyzed and configured in the software according to Chapter 3. The seven L/M-pairs then focus the configured sound data to the L/M that sent the sound data, i.e. the point of focus. The remaining L/M of the eighth L/M-pair is then used to verify that the signal energy is suppressed in areas around the point of focus. That is, the user gets information concerning the power of the signal in the point of focus and in a point around the point of focus, therefore being able to verify that the signal energy is focused in the point of focus and suppressed around the point of focus. Thereby, some of the powers of massive MIMO beamforming is shown.

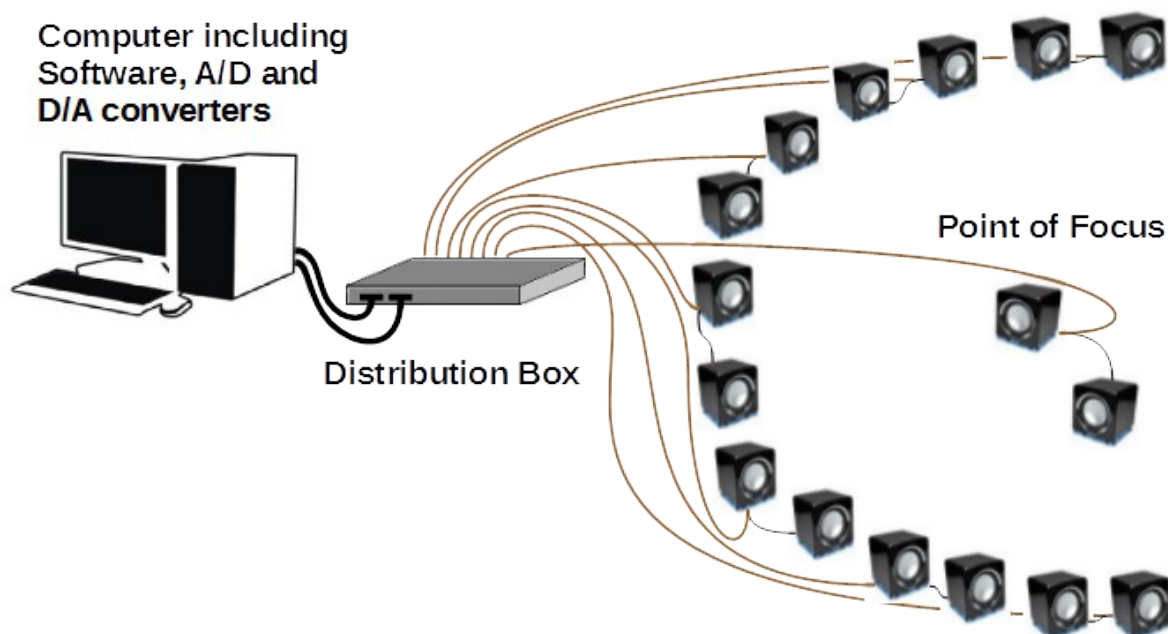


Figure 1: An overview of the system demonstrating that signal energy can be focused into one precise point and at the same time be suppressed essentially everywhere else.



3 SOFTWARE

The software consists of a user interface, an OS, four software modules and also drivers for the A/D- and D/A converters. The modules are written in MATLAB and are used to take care of all the computations and will also provide the interface for the user towards the hardware. The modules and the other parts of the software are described in Sections 3.2 to 3.7. For an overview of the software see Fig. 2.

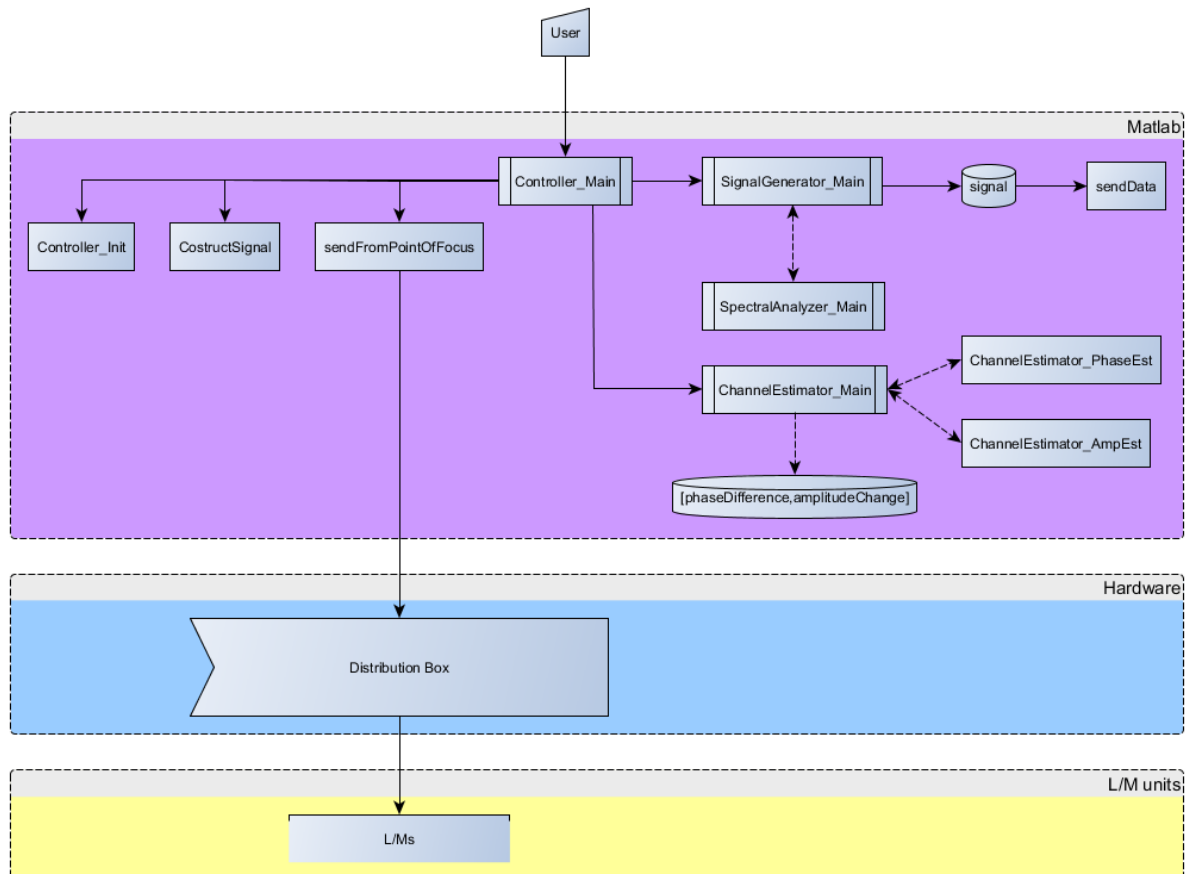


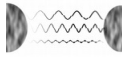
Figure 2: An overview of the software.

3.1 General Implementation of Modules

Everything belonging to the different modules (see Sections 3.3 to 3.6) is MATLAB code in the form of functions or scripts. Both the scripts and the functions have a description at the top describing what they do.

An important difference between scripts and functions are what variables they can access. Scripts share all variables with the calling instance, be it a function or a script, and any variable defined in a script will also be available after the script has returned in the caller of the script. Functions on the other hand have access to no variables whatsoever except for those passed along through its input parameters, and of all the variables defined in the function only the ones specified as out parameters are available to the caller.

By default, MATLAB stores all numeric variables as double-precision floating-point values. Unless otherwise specified, all output and input to functions are of this data type.



3.2 User Interface

The User Interface consists of the MATLAB command line interface. The user interacts with the system by calling the `Controller_Main` script, which in turn calls other scripts and functions.

3.3 Controller

The Controller is the main module coordinating the modules described in Sections 3.4, 3.5 and 3.6. The main purpose of this module is to centralize the communication between the different modules in order to maintain a manageable structure and enforce loose coupling between the different modules. The Controller is the users only access point to the systems software.

The Controller also handles the communication towards the hardware. It is responsible for ensuring that all signals are transmitted synchronously and with the intended phase and amplitude.

3.3.1 Controller_Main

The purpose of this script is to serve as a main loop that calls on all other scripts and functions, when appropriate, to run a complete demonstration of the system.

`Controller_Main` first calls the script `Controller_Init`. Once that has been run, all the central variables have been initialized and set. The pilot is then created through a call to `constructSignal`, after which the pilot is used as input to the function `sendFromPointOfFocus`. The returned data, a matrix containing the data received at the peripheral L/M-units, is then kept for future use.

In the next step the script loops over each of the peripheral L/M-units, as most of the following functions take data corresponding to one L/M as input. First `ChannelEstimator_Main` is called, using the pilot together with one column of the previously recorded data as primary input. Once that has been called for all peripheral L/M-units, the function `SignalGenerator_ModAmp` is called using all amplitude change factors at once. `Controller_Main` then proceeds to once again loop over all the peripheral L/M-units. With the received, and altered channel information, `SignalGenerator_Main` is called, returning a suitable signal to transmit from the relevant L/M-unit. After this step the second loop ends.

Now that the loop has completed, all necessary knowledge is available to complete the final step of the demonstration. Using the newly constructed signals, that are phase shifted and scaled version of the pilot, the function `sendData` is called. Once the L/M-units have finished transmitting, the recorded signals from the point of focus and the point of reference are returned.

Depending on the users choice of settings, some data is displayed, presenting the result. The default is to show a plot of the two recorded signals in the time domain as well as periodograms of the two signals. The periodograms are attained through calls to `SpectralAnalyzer_Main`. The difference in power in dB is also printed. A positive value indicates that the sound recorded at the point of focus was stronger than the sound recorded at the point of reference.

The script then proceeds with either restarting from the beginning or ending.

3.3.2 Controller_Init

The purpose of this script is to initialize the variables that specify how the system is run. For most variables the user can choose to either use a default value or to set a new value to use for this run of the system. Important variables and constants found in this script are presented in Table 1.

The script starts by defining the different variables that the user may choose to alter, as well as a few constants that are used as limits for what values may be used. Using loops, error checks and prompts the script then asks the user whether the variables should be set manually or not, potentially followed by several prompts regarding the different variables. Finally the analog output and input variables are created and set to their initial settings.

*Table 1: Variables and constants in the Controller.*

Variables	Definition
amountLMPairs	The total amount of L/M-pairs used in the system
amountLMGroups	The amount of groups of L/Ms used in the system
groupAtFocus	The control group that is used as the point of focus {Introduce some definition}
carrierFreq	The carrier frequency of the signal that is to be constructed
timeDuration	The time duration of the signal that is to be constructed
masterPoF	The L/M-unit in the L/M-pair at the point of focus that functions as the actual point of focus, 1 for master 0 for slave
LMFocusPair	The pair (a number) that is used as point of focus (and is subsequently part of the group specified by groupAtFocus)
whatToPlot	A value that plots and prints different information depending on its value
seeEmittedChars	A value that decides whether the changes in phase taken into account when using function sendData should be shown or not
Constants	Definition
sampleFreq	The sample frequency that is used throughout the system
carrierFreqMin	The lower limit on carrier frequencies
carrierFreqMax	The upper limit on carrier frequencies
timeDurationMax	The upper limit on time duration
sampleFreqMax	The maximal sampling frequency, bounded by hardware constant divided by amount of channels in use

3.3.3 constructSignal

The purpose of this function is to construct the signal that is to be used when transmitting from the L/M-unit at the point of focus as well as from the peripheral L/M-units. All in and out parameters are presented in Table 2.

Using the available input parameters the function constructs a vector corresponding to a sine with the correct frequency, duration and sample time.

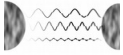
Table 2: In and out parameters in constructSignal

In parameters	Definition
carrierFreq	The carrier frequency of the signal that is to be constructed
timeDuration	The time duration of the signal that is to be constructed
sampleFreq	The sample frequency that is used throughout the system
Out parameters	Definition
pilot	The pilot that is to be used throughout the system to measure the channel as well as producing the MIMO effect

3.3.4 sendFromPointOfFocus

The purpose of this function is to transmit a known pilot from the L/M-unit being used as point of focus while recording at all the peripheral L/M-units, and then save this data so that estimations of the effects of the channel on phase and amplitude can be done. All in and out parameters are presented in Table 3.

The function starts by changing the values of the digital output lines in such a way that all the L/M-pairs at the group at focus are set to loudspeaker mode while the rest of the L/M-pairs are set to



microphone mode. The analog output channel is then set to accept values from zero to plus ten volts and the analog input channel is set to accept values from minus ten to plus ten volts.

After this the transmitting L/M-unit transmits the actual pilot for one second before it stops. If the transmitting L/M-unit is the slave of a pair the amplitude of the outgoing signal is amplified by a factor of two. Meanwhile, the peripheral L/M-units are recording. The sound vectors recorded are then combined into a matrix and returned.

Table 3: In and out parameters in sendFromPointOfFocus.

In parameters	Definition
pilot	The signal that is to be transmitted from the point of focus and then used as reference when estimating the channel
ao	The analog output object used to interact with the D/A-card
ai	The analog input object used to interact with the A/D-card
amountLMPairs	The number of L/M-pairs currently in use, including pair at the point of focus {Introduce some definition}
masterPoF	The L/M-unit in the L/M-pair at the point of focus that functions as the actual point of focus, 1 for master 0 for slave {Introduce some definitions}
groupAtFocus	The control group that is used as the point of focus {Introduce some definition}
LMFocusPair	The pair (a number) that is used as point of focus (and is subsequently part of the group specified by groupAtFocus)
sampleFreq	The sampling frequency used for both the A/D- and D/A-card
Out parameters	Definition
PointOfFocus	The time domain signal received at the point of focus
verifier	The time domain signal received at the verifier

3.3.5 sendData

The purpose of sendData is to synchronously transmit from the peripheral L/M-units and receive at the L/M-pair at the point of focus. This function is to be executed after having measured the channel through other means, since only then will we be able to correctly alter the phase and amplitude of the pilot correctly with respect to the situation of each individual L/M-unit. All in and out parameters are presented in Table 4.

The function starts by changing the values of the digital output lines in such a way that all the L/M-pairs in the group at focus are set to microphone mode while the rest of the L/M-pairs are set to loudspeaker mode. The analog output channel is then set to accept values from zero to plus ten volts and the analog input channel is set to accept values from minus ten to plus ten volts.

Depending on the value of the timeDuration variable, two different methods are then used for queuing the sound data on the D/A-card. This is due to there being a limit to how much data the D/A-card can process at a time. The threshold is set to two seconds. If the function is to transmit for longer than this time period the function uses a different trigger setting combined with a loop that continuously puts new data in the output buffer. For shorter time periods than two seconds the buffer can fit all the data needed and need thus not be updated while being run.

Regardless of the method used to transmit, the A/D-card records data for exactly one second from the two L/M-units belonging to the L/M-pair specified by the variable LMFocusPair. The data recorded in the slave of the pair at the point of focus is then scaled by 2 dB to even out some hardware differences. The two recorded vectors are then returned.

*Table 4: In and out parameters in sendData.*

In parameters	Definition
signal_vector	A matrix that contains the time domain signals that are to be sent from the L/M-units
ao	The analog output object used to interact with the D/A-card
ai	The analog input object used to interact with the A/D-card
amountLMPairs	The number of L/M-pairs currently in use, including pair at the point of focus {Introduce some definition}
timeDuration	The time in seconds that the peripheral L/M-units are to transmit sound
sampleFreq	The sampling frequency used for both the A/D- and D/A-card
masterPoF	The L/M-unit in the L/M-pair at the point of focus that functions as the actual point of focus, 1 for master 0 for slave {Introduce some definitions}
groupAtFocus	The control group that is used as the point of focus {Introduce some definition}
LMFocusPair	The pair (a number) that is used as point of focus (and is subsequently part of the group specified by groupAtFocus)
Out parameters	Definition
PointOfFocus	The time domain signal received at the point of focus
verifier	The time domain signal received at the verifier

3.4 Channel Estimator

The Channel Estimator is a function with the intended purpose of measuring the channel properties of an individual L/M, based upon the sound data measured by the L/M. The Channel Estimator returns information about the effects of the channel on the amplitude and the phase of the sent sound. This is done by comparing the known signal sent for this purpose with the same signal after having passed through the channel.

In order for this to be effective, the noise level has to be minimal meaning that as many noise sources as possible have to be eliminated. The L/Ms can therefore be placed on an elastic surface in order to reduce vibrations that possibly would affect the measurements. The circuits within the Distribution Box are also isolated by the enclosure, in order to avoid electromagnetic interference.

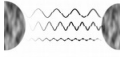
3.4.1 ChannelEstimator_Main

The purpose of ChannelEstimator_Main is to act as an intermediary between the Controller and all the other sub functions belonging to the Channel Estimator. All in and out parameters are presented in Table 5.

ChannelEstimator_Main simply calls the two functions ChannelEstimator_AmpEst and ChannelEstimator_PhaseEst. After that it determines what to plot; if whatToPlot has a value of one, two or three the function plots the original signal, the altered signal or both. If whatToPlot has any other value nothing is plotted. The function then returns.

Table 5: In and out parameters in ChannelEstimator_Main.

In parameters	Definition
originalSound	The signal that is used as reference when computing the differences
alteredSound	The signal that was measured after originalSignal was sent through a channel of some kind
sampleFreq	The sample frequency that is used throughout the system
whatToPlot	A value that determines whether the time domain signals of originalSignal and/or alteredSignal



Out parameters	Definition
amplitudeChange	The average change in amplitude between originalSignal and alteredSignal
phaseDifference	The difference in phase between originalSignal and alteredSignal

3.4.2 ChannelEstimator_AmpEst

The purpose of this function is to calculate the change in amplitude imposed when a signal passes through a channel and is recorded by a specific L/M-unit. It works under the assumption that the change in amplitude is approximately constant over the time period of the signal. All in and out parameters are presented in Table 6.

ChannelEstimator_Main works by first dividing originalSignal into a hundred equally long parts. For each part the maximum value is computed, resulting in a hundred different scalars. The mean value of these scalars is then represents the mean amplitude of this signal.

The same operations are done for alteredSignal, resulting in another mean value. To get the actual difference in amplitude as a factor the two values are divided with the mean value from alteredSignal as numerator. This value is then returned.

Table 6: In and out parameters in ChannelEstimator_AmpEst.

In parameters	Definition
originalSound	The signal that is used as reference when computing the difference in amplitude
alteredSound	The signal that was measured after originalSignal was sent through a channel of some kind
Out parameters	Definition
amplitudeChange	The average change in amplitude between originalSignal and alteredSignal

3.4.3 ChannelEstimator_PhaseEst

This function calculates the difference in phase between two input signals and returns it. All in and out parameters are presented in Table 7.

The phase difference is found using MATLAB's implementation of the cross correlation function according to the equations

$$i = \underset{m}{\operatorname{argmax}} \begin{cases} \sum_{n=0}^{N-m-1} x[n+m] * y^*[n], & m \geq 0 \\ \sum_{n=0}^{N+m-1} x^*[n-m] * y[n], & m < 0 \end{cases}$$

$$t = \frac{i}{\text{sampleFreq}}$$

$$\text{phaseDifference} = \operatorname{mod}(t * f * 2 * \pi, 2 * \pi)$$

where x and y are the two input signals, N is the length of the input signals and f is the carrier frequency of both input signals [2].

Table 7: In and out parameters in ChannelEstimator_PhaseEst.

In parameters	Definition
originalSound	The signal that is used as reference when computing the difference in phase
alteredSound	The signal that was measured after originalSignal was sent through a channel of some kind
sampleFreq	The sample frequency that is used throughout the system



Out parameters	Definition
phaseDifference	The difference in phase between the two input signals

3.5 Spectral Analyzer

The primary purpose of the Spectral Analyzer, which consists of only one function, is to calculate the main frequency component of a signal. In addition to this, it is also able to plot the frequency spectrum as well as a periodogram of an input signal.

3.5.1 SpectralAnalyzer_Main

The function first computes the discrete Fourier transform (DFT) of the variable signal. The index of the maximum value of the DFT then corresponds to the main frequency component. This value is found through MATLAB's find function. Using sampleFreq this index is thus translated to a frequency. All in and out parameters are presented in Table 8.

The variable whatToPlot then determines whether the Spectral Analyzer also plots the frequency spectrum and/or the periodogram. The frequency spectrum was calculated when taking the DFT of the input signal. The periodogram, which is an estimate of the spectral density function, is acquired by first dividing the signal into five equally large parts. The DFT of these parts are computed and then squared. Finally the mean is computed. The plotting itself is done with the available plot function in MATLAB.

Table 8: In and out parameters in SpectralAnalyzer_Main.

In parameters	Definition
signal	The signal that is to be analyzed
sampleFreq	The sampling frequency that is used throughout the system
whatToPlot	A value that determines what to plot
Out parameters	Definition
freq	The average change in amplitude between originalSignal and alteredSignal

3.6 Signal Generator

The purpose of the Signal Generator module is to generate the signals that are to be transmitted towards the point of focus. This includes scaling the signals as well as altering their phase.

3.6.1 SignalGenerator_Main

The purpose of this function is to alter a given signal so that it perfectly counters the amplitude change and the phase change. All in and out parameters are presented in Table 9.

The scaling of the amplitude is performed by simply multiplying desiredSignal with absAmpDiff. To change the phase phaseToShift, which is a phase in radians, it is first transformed into an offset counted in index steps through use of the sampling frequency, sampleFreq. The shift in the signal is then realized through a shortening of the desiredSignal. The first few samples, equal to the calculated index offset in length, is removed. This results in a negative shift of size phaseToShift meaning that outputSignal would ideally end up with a net phase shift of zero if passed through the channel again. It is assumed that the phase sent to this function, phaseToShift, is positive.

Table 9: In and out parameters in SignalGenerator_Main.

In parameters	Definition
desiredSignal	The signal that is to be analyzed
absAmpDiff	The factor that desiredSignal is to be scaled with to counter the channels effects
phaseToShift	The phase that is to be countered
sampleFreq	The sampling frequency that is used throughout the system



Out parameters	Definition
outputSignal	The output signal in the form of a vector

3.6.2 SignalGenerator_ModAmp

The purpose of SignalGenerator_ModAmp is to function as an intermediary step between the measurement of the amplitude difference and the usage of this information in SignalGenerator_Main. The idea is to look at the amplitude changes of all the L/M-units at once to perform a relative scaling, which is impossible to do in SignalGenerator_Main, with the goal of ensuring that the voltage levels will not be too high. This is necessary to do in a separate function as SignalGenerator_Main only considers one signal at a time. The output is the factor that should then be multiplied with the signal in SignalGenerator_Main. All in and out parameters are presented in Table 10.

All elements in ampDiffs are first inverted, resulting in appropriate factors if the only goal was to counter the channels for signals individually. All of these factors are then divided by the maximum of the factors, and then multiplied by 0.5. The division with the maximum ensures that the relative quotients remain constant while also scaling the values so that the maximum now is one. The last factor, 0.5, is simply to further dampen the output voltage as the highest voltage output that we want to generate at this stage is 0.5 volts.

Table 10: In and out parameters in SignalGenerator_ModAmp.

In parameters	Definition
ampDiffs	The change in amplitude imposed by all of the measured channels
Out parameters	Definition
absAmpDiffs	The factors to use in SignalGenerator_Main for amplitude scaling

3.7 OS/Drivers

The OS and the drivers constitute the interface between the MATLAB code and the hardware. The installed OS is Microsoft Windows 7. The different drivers are the drivers for the A/D and D/A converters. Additionally, there is also an add-on for MATLAB called Data Acquisition Toolbox (DAQ) as well as a MATLAB library called MATLAB-compliant Data Acquisition Library (MLDAQ) installed.

4 HARDWARE

The hardware consists of a computer, an A/D converter, a D/A converter, a distribution box and L/M-pairs. All these entities are described in this chapter.

4.1 Computer

The computer is the platform where all the software is operating. The computer that is included in the product is a desktop computer from HP, more specifically a HP Compaq Elite 8300 with Windows 7 as OS. There is one A/D and one D/A converter installed into the PCI slots on the motherboard of the computer.

4.2 A/D and D/A Converters

The A/D and D/A converters are, as stated in Section 4.1, installed into the PCI slots of the computer and serve as an intermediate communication link between the computer and the distribution box in the overall channel between the computer and the L/M-pairs. There is one A/D and one D/A converter. The A/D converter is of type Contec AD12-64 (PCI) and the D/A converter is of type Contec DA12-16 (PCI).



Both converters have a resolution of 12 bits and a maximum conversion speed of 100 kilosamples/s, which can be achieved with the internal sampling clock of each card. The output voltage levels from the D/A converter and also the input voltage levels of the A/D converter are both restricted to the interval $[-10, 10]$ V, however they can be further divided into smaller intervals in order to gain precision.

Besides having 64 analog input channels, the A/D converter also has four digital outputs and four digital inputs for TTL level signals (Transistor-Transistor-Logic). In the product, only two of the digital outputs are used. These outputs are used to switch between the operational modes of the L/M-pairs. The A/D converter has 16 output channels, which are all used in the product. A more detailed description of the connections between the converters and distribution box is given in Section 4.4.1.

For more details about the A/D- and D/A converters, see the web page of the manufacturer, Contec [3].

4.3 L/M

An L/M-pair consists of two loudspeakers, one master unit and one slave unit. The master unit consists of an original amplifier board and an additional detection board, designed by Mikael Olofsson. The detection board enables the L/M-pair to have two modes of operation, namely loudspeaker mode and microphone mode. The interface, for signals sent to and from the L/M-pair, is through a 9-pole D-sub connector (DB-9), mounted on top of the master unit. The power supply for an L/M-pair is through a USB cable, connected to a wall socket via a USB adapter. The following sections, Section 4.3.1 and 4.3.2, describe in detail what components are used and how the components are put together to create an L/M-pair.

4.3.1 Detection Board

The detection board is the circuit that enables the microphone mode for the L/M-pair. When the L/M-pair is working in microphone mode, the circuit amplifies the audio signals received using a differential-in-differential-out amplifier with a voltage gain of 23 dB. Then the amplified signals are sent to the collection board for further amplification, see Section 4.4.2.

In order to create an L/M-pair, a detection board is added to the master loudspeaker. The detection board consists of a copper breadboard, electrical components and cables, see Fig. 3. The components and the cables are soldered to the copper breadboard. The components used for one detection board are listed in Table 11. Note that the socket is not required for the detection board to work, but was included to cope with the fact that the electronics supplier, Elfa, was out of stock of an operational amplifier and because of this there was a delay of the delivery of this operational amplifier. The amplifiers could then quickly be inserted upon arrival, without soldering. Also two different types of operation amplifiers are used for the same reason.

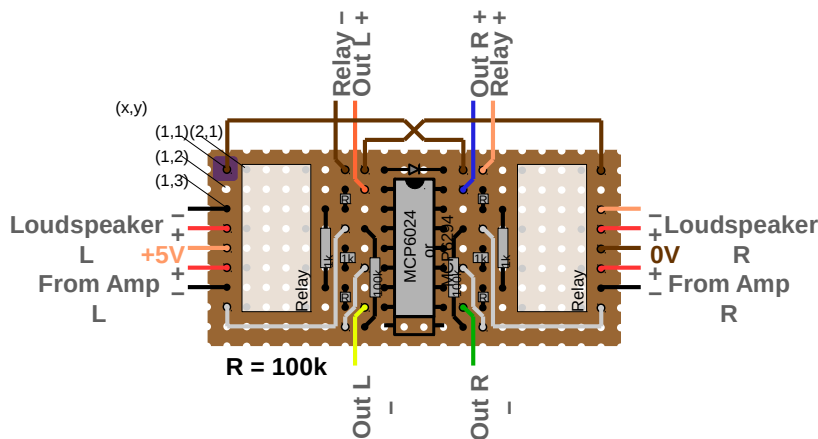


Figure 3: Detection board from above, plastic side facing up.

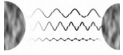


Table 11: Listing of components needed for one detection board.

Component type	Value	Product number	# of units needed	Stock number (ELFA)
Resistor	1k Ohm	-	4	-
Resistor	100k Ohm	-	6	-
Relay	5VDC, 167 Ohm	HFD27F/005-H 051	2	37-157-20
Socket	16 pin	-	0-1	-
OP-Amp	Quad	MCP6024-I/P or MCP6294-E/P	1	73-786-63 or 73-786-92
Capacitor	1 μ F	-	1	-
Diode	75V	1N914	1	70-190-50

The copper breadboard has a dimension of nine times twenty holes and the copper lines are aligned such that there are nine copper lines with twenty holes in each line. However, the copper lines are cut at some specific places, indicated as crosses in Fig. 4, and also some bridges between power lines are created to provide the correct connections, see Fig. 4.

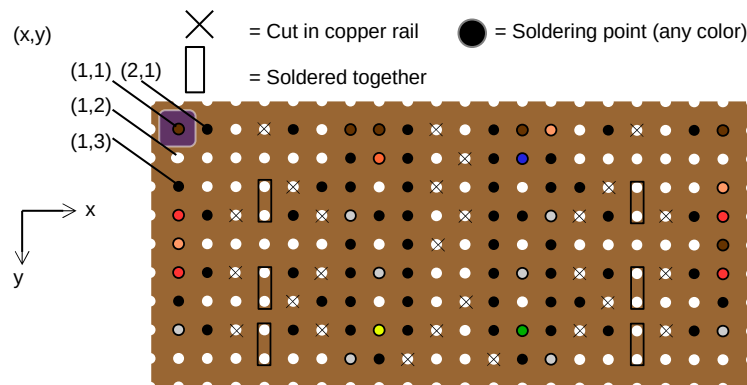


Figure 4: Detection board from above, without components. Figure shows where copper rail on solder side is to be cut or connected.

As seen in Fig. 3 and Fig. 4, there are small markings in the top left corner. These markings indicate the origin for a coordinate system, where x increases to the right and y increases downwards. In Table 12, the cables from the detection board to the master loudspeaker and the DB-9 are specified.

Table 12: Listing of external cables originating from the detection board.

From detection board (x,y), plastic side facing upwards	To connection	Color of cable	Length of cable
(1,4)	Master loudspeaker +	Red	12 cm
(1,3)	Master loudspeaker -	Black	12 cm
(8,8)	DB-9 pin 6	Yellow	17 cm
(14,1)	DB-9 pin 7	Light brown	17 cm
(13,2)	DB-9 pin 5	Blue	17 cm
(8,2)	DB-9 pin 1	Orange	17 cm
(13,8)	DB-9 pin 9	Green	17 cm



From detection board (x,y), plastic side facing upwards	To connection	Color of cable	Length of cable
(7,1)	DB-9 pin 8	Brown	17 cm

On the soldering side of the copper breadboard, the capacitor is mounted. This is done after all solderings are finished, even the ones between the original circuit and the detection board. The solderings between the original board and the detection board are presented in Section 4.3.2. The capacitor is provided a piece of heat shrink tubing on the longest leg to minimize the risk of unwanted connections. The capacitor is connected with the short leg (minus) to (5,5) and the long leg to (10,6), the coordinates are valid when the copper side is facing upwards.

4.3.2 Maxxtro Mini Speaker 4 W

The base for an L/M-pair is the “Maxxtro mini speaker 4 W”, consisting of a master loudspeaker and a slave loudspeaker. The master unit is the loudspeaker that has the volume control, the USB cable and the 3.5 mm cable connected to it. The loudspeakers are active and driven by 5V DC, provided by the USB cable. The USB cable is connected to an adapter named “Euro-USB-laddare”, which has the stock number 25-249-98 at ELFA.

The L/M-pair is constructed by modifying the master unit of the loudspeaker pair. That is, adding a detection board and a DB-9 to the master unit, as well as a lot of cable connections. To accomplish this, the master unit is disassembled by removing four screws at the backside of the loudspeaker enclosure. When all modifications are carried out, the master unit is assembled again. The following paragraphs will explain what modifications are carried out.

Inside the master unit, an original circuit is found, which requires some desoldering and soldering in order to create the functionality required for an L/M-pair. The connections for the loudspeaker, the light emitting diode and the cable from the slave unit, consisting of four thin cables (blue, green, red, copper) soldered onto the soldering side of the original board, the diode gets new red cables of length 7 cm, together with heat shrink tubing close to the diode, and is soldered to “LED2” on the original board, from the side opposite of the soldering side. The thin blue and green cables from the slave unit is soldered to “LED1” on the original board, with the blue cable fastened to the anode side of the diode symbol. The original board can be seen in Fig. 5.

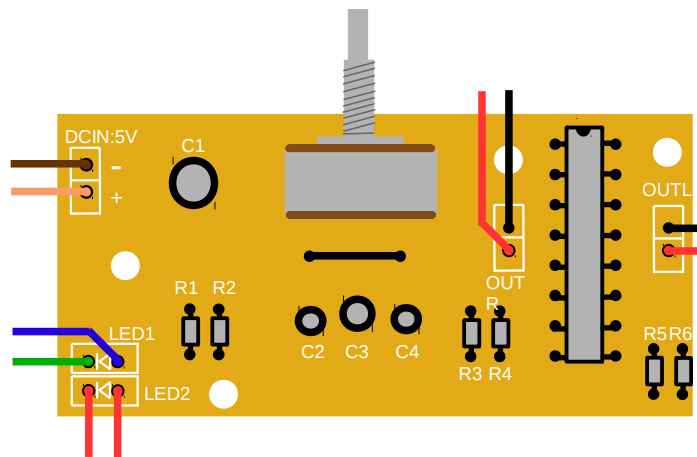
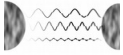


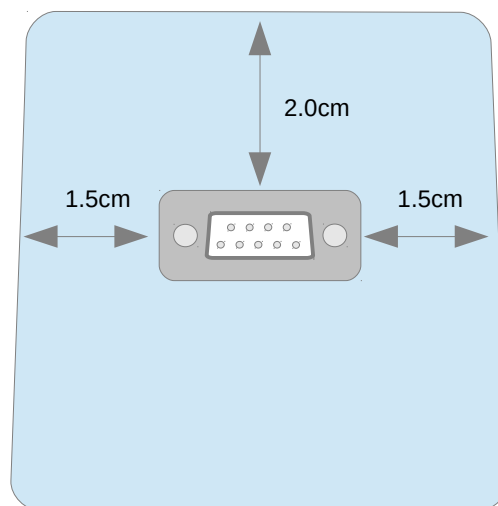
Figure 5: Original board from above, soldering side facing down.

The thin red and copper colored cables from the slave unit are soldered to the detection board at coordinates (20,3), for the copper colored cable, and (20,4), for the red cable, from the plastic side of the detection board. The following table, table 4, specifies the rest of the cables originating from the original board. They are connected either to the detection board or to the DB-9.

*Table 13: Listing of external cables originating from the original board.*

From original board	To connection	Color of cable	Length of cable
DCIN 5V +	Detection board (1,5)	Light brown	10 cm
DCIN 5V –	Detection board (20,5)	Dark brown	7 cm
Out L +	Detection board (1,6)	Red	8 cm
Out L –	Detection board (1,7)	Black	8 cm
Out R +	Detection board (20,6)	Red	8 cm
Out R –	Detection board (20,7)	Black	8 cm
G	DB-9 pin 3	Black	17 cm
R	DB-9 pin 4	Gray	17 cm
L	DB-9 pin 2	Purple	17 cm

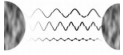
The DB-9 is mounted to the top of the master unit and therefore three holes are drilled, one common hole for all pins and two holes for the screws. Fig. 6 shows how the DB-9 is mounted when the required holes have been drilled.

*Figure 6: Placement of DB-9 on top of master unit in L/M-pair.*

The cables from the original board and the copper The cables from the original board and the copper breadboard are soldered to the DB-9 and then the DB-9 is mounted on the top of the master unit. The DB-9 is fastened with a distance, a round plate, a spring plate and a screw nut, in that specific order. Table 5 lists the components used to fasten the DB-9.

Table 14: Listing of components used to fasten one DB-9.

Component type	Product number	# of units needed	Stock number (ELFA)
Distance	99044-0003-8 / D-SUB DISTM	2	48-006-85
Round plate	BRB 3,2X7X0,5 / CLL726 3,2	2	48-000-19
Spring plate	FBB 3,1X6,2X0,8 / CLL1261	2	48-000-42
Screw nut	M6MS M3 / CLL629 MUTTER A4	2	48-000-81



The black and red cables for the master loudspeaker, from detection board, are soldered to the master loudspeaker and the both circuits are put inside the enclosure according to Fig. 7. There is a piece of paper in between the circuits to prevent unwanted connections. The light emitting diode is put into the hole in the plastic bottom and the front panel is put back to the enclosure, finishing the modifications required to get an L/M-pair.

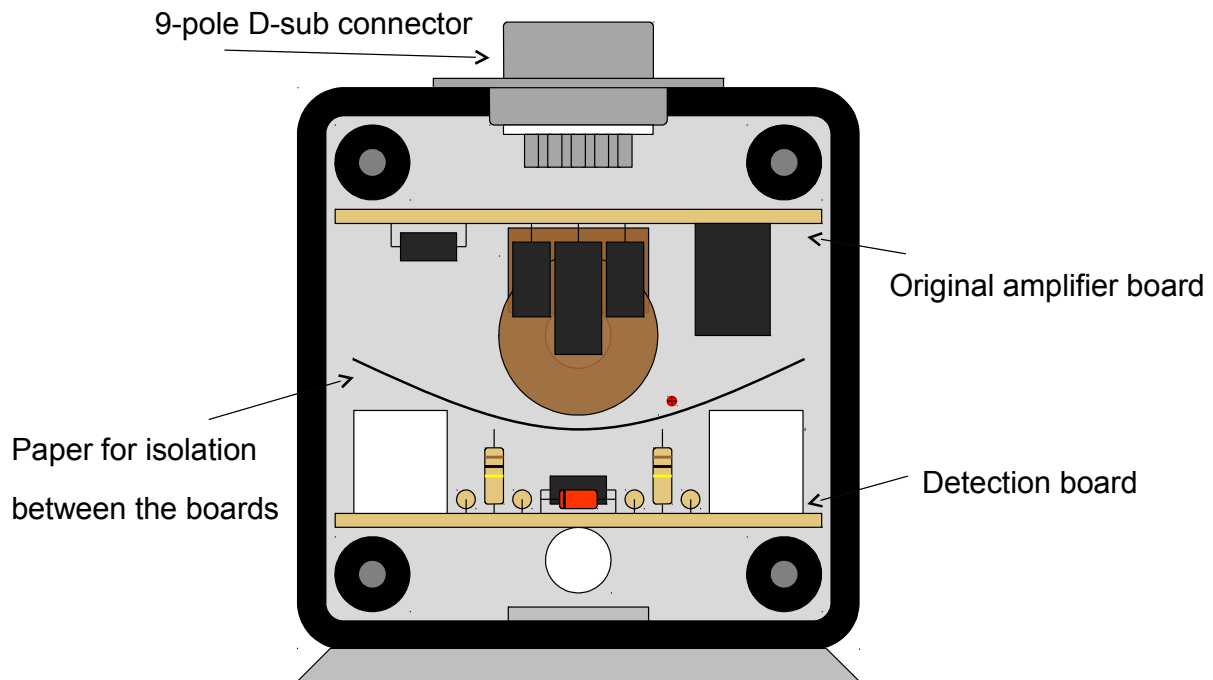


Figure 7: Master unit of L/M-pair populated with original board and detection board.

4.4 Distribution Box

The purpose of the distribution box is to distribute the signals, from and to the computer and the loudspeaker units. It works as a hub and hence canalizes the signals to the correct locations. It has a total of 11 connections. Eight for the L/M-pairs, one each for the A/D and D/A converters respectively and one connection for power supply. The distribution box also has a control board consisting of eight two-way switches, with the purpose of enabling the user to choose which L/M-pair is connected to which control group. For an overview of the distribution box, see Fig. 8. The front panel, where all the DB9 contacts are placed, is shown in Fig. 9.

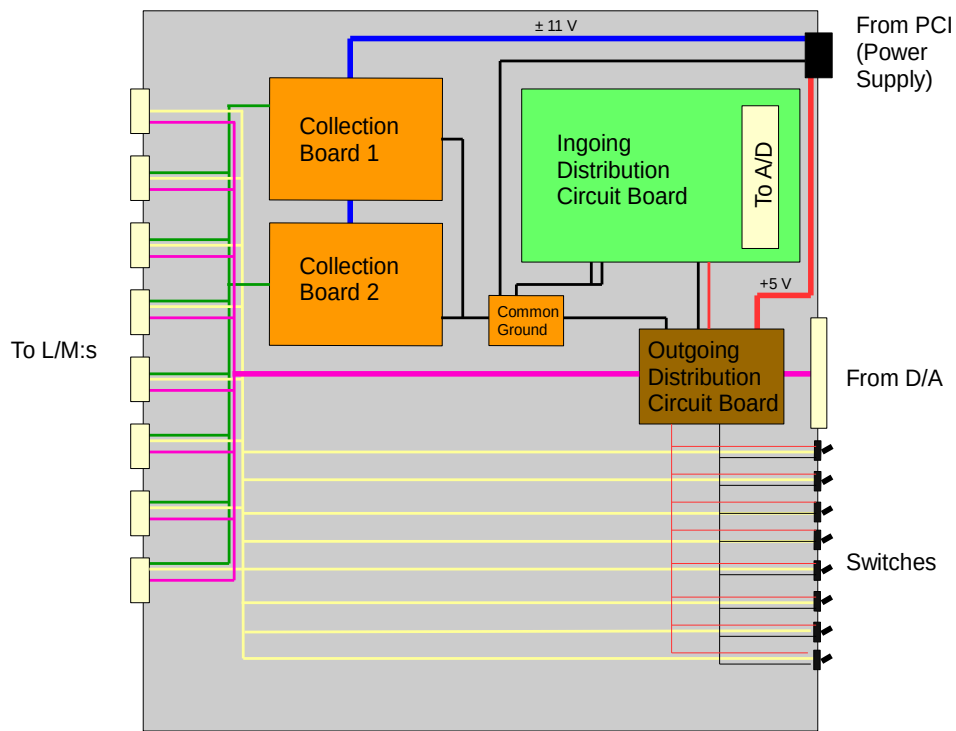
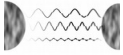


Figure 8: Overview of the Distribution Box. The figure is a simple layout, which omits the actual cables. The purpose is to only show the connections between the circuits.

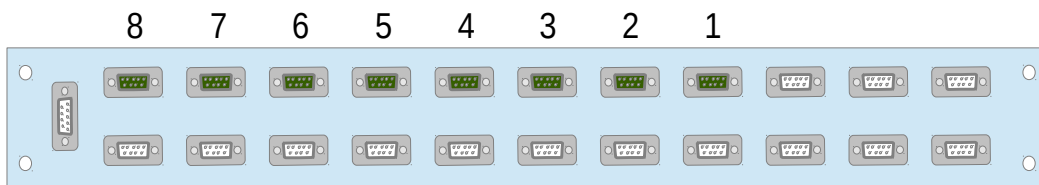


Figure 9: Layout of the front panel, showing what DB-9 connectors that are used and which L/M-units they connect to.

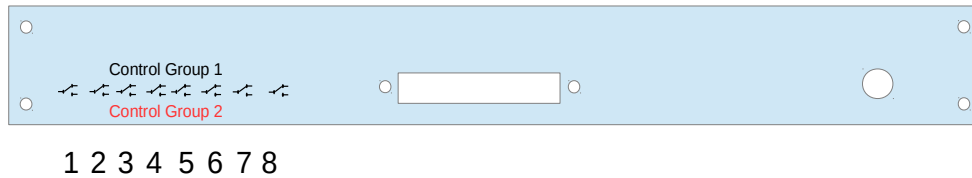


Figure 10: Layout of the back panel. The picture shows the placing of the control switches, the DB-37 connector and the hole for the power supply.

4.4.1 Outgoing Distribution Circuit Board

The Outgoing Distribution Circuit Board is mainly a connection between the DB-37 connection from the D/A converter and the L/M-pairs, in order to keep track of which speaker corresponds to which port in the D/A converter. The board does although also have a circuit for the relay controls, which consists of 4 transistors, 2 for each control group, and 2 resistors. The purpose of this circuit is to limit the current coming from the A/D card, and still supply the relays in the L/M-pairs with the appropriate voltage of 5 volts in order to switch them into microphone mode when desired. The software controlled binary signal is placed upon the base of the smaller transistor, which then in turn is connected to the base of the bigger transistor. When a signal is generated from the A/D card, the second transistor is then opened and connects the negative pole of the relays to ground, thus creating a voltage on the relays and switching the function of the L/M-pair. Which of the control groups the L/M-pair is connected to is controlled by a mechanical switch, mounted on the back panel of the distribution box, as seen in Fig. 10.

The layout of the Outgoing Distribution Circuit Board is shown in Fig. 11 and Fig. 12.

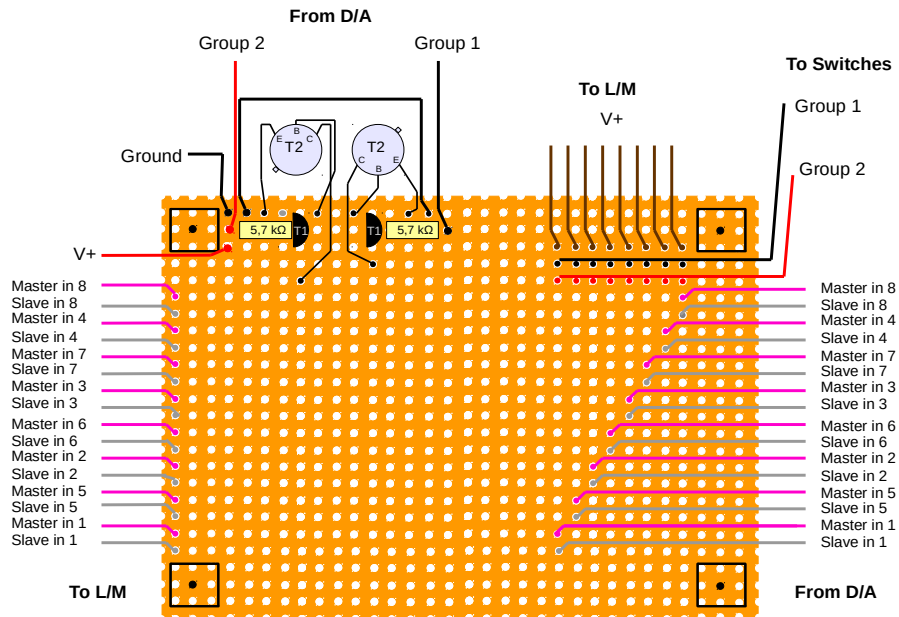
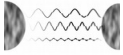


Figure 11: Layout of the Outgoing Distribution Circuit Board. The figure shows the board from the component side, with the components and the connecting cables.

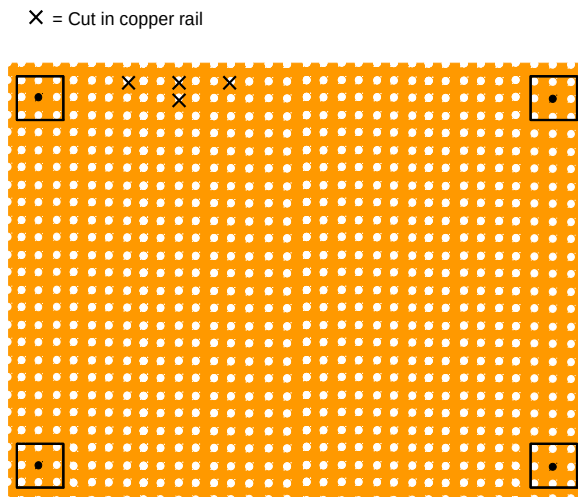


Figure 12: Cuttings on the solder side. The figure shows the board from the component side, without the components.



Table 15: Listing of components required for one relay control board.

Component type	Value	Product number	# of units needed	Stock number (ELFA)
Resistor	5.6k Ohm	-	2	-
Transistor	gain = 250	BC337-40	2	71-014-59
Transistor	gain = 15	BSX47-10-T	2	71-301-48

The corner rectangles in both pictures represent the holes to be drilled for mounting purposes. It is important to make sure that the screws are isolated from the copper rails on the solder side, or else the corresponding rails will be grounded to the common ground of the cabinet of the distribution box, which will lead to the circuit not working correctly. The isolation is done by cutting the copper rails.

4.4.2 Collection Board

The collection board constitutes the second stage of amplification, and amplifies the signals that is received from the detection board of the L/M-pairs. There are a total of two equal circuits, each consisting of eight amplification circuits, two for each L/M-pair. Each amplification circuit takes in a differential signal and amplifies the difference. The layout of the circuit board is shown in Fig. 13 and Fig. 14.

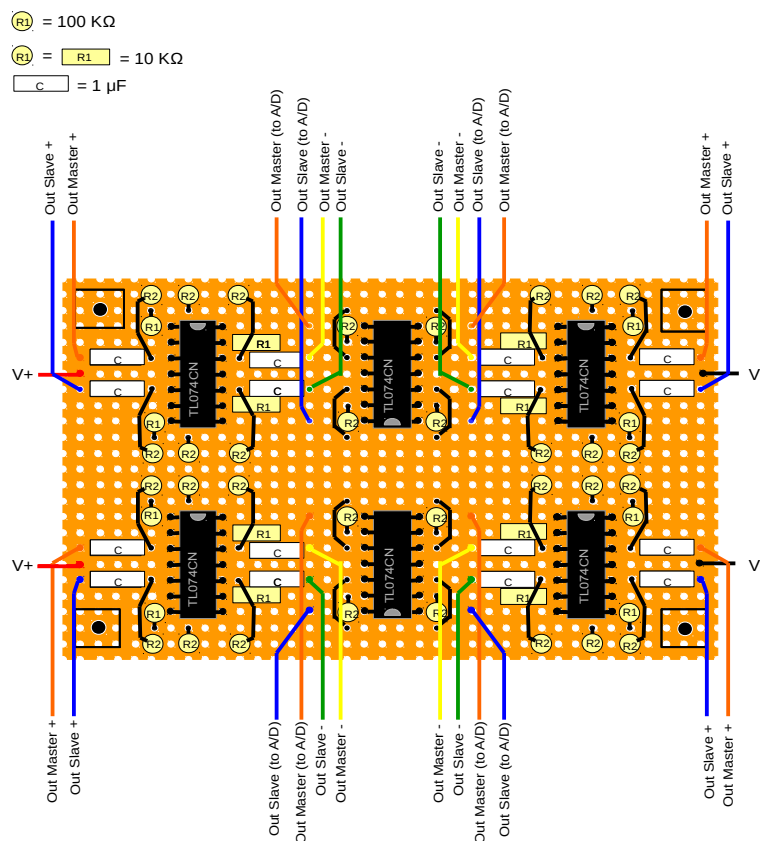


Figure 13: The figure shows one of the two collection board circuits seen from the component side, complete with all components.

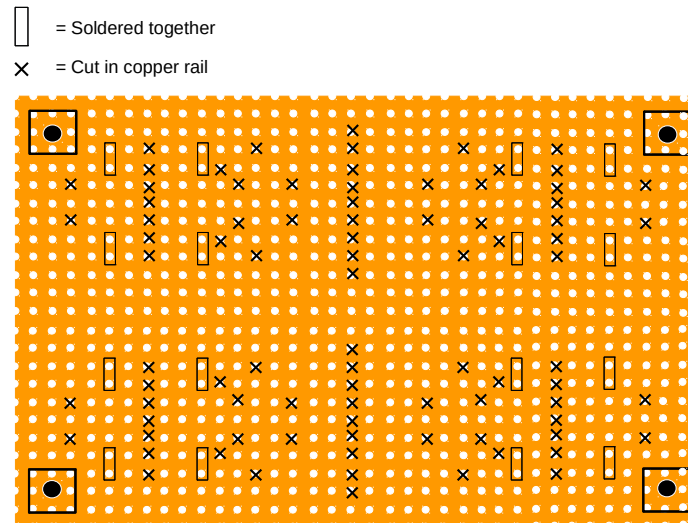
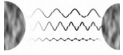


Figure 14: Collection board seen from the component side. The figure shows where the copper rails need to be cut or soldered together.

Again the corner rectangles represent the screw holes for mounting purposes. It is also in this case important to isolate the copper rails from the screws.

Table 16: Listing of components required for one collection board.

Component type	Value	Product number	# of units needed	Stock number (ELFA)
Resistor	10k Ohm	-	32	-
Resistor	100k Ohm	-	64	-
Capacitor	1 μ F	-	32	-
OP-Amp	Quad	TL074CN	12	73-117-56

4.4.3 Ingoing Distribution Circuit Board

The Ingoing Distribution Circuit Board, see Fig. 15, is where the output of the collection board is channeled into. It consists of a Contec DTP-64 screw terminal, which allows for connecting cables without soldering. The screw terminal is mounted inside the distribution box and connected to the A/D converter via a PCA cable with 96 pins.

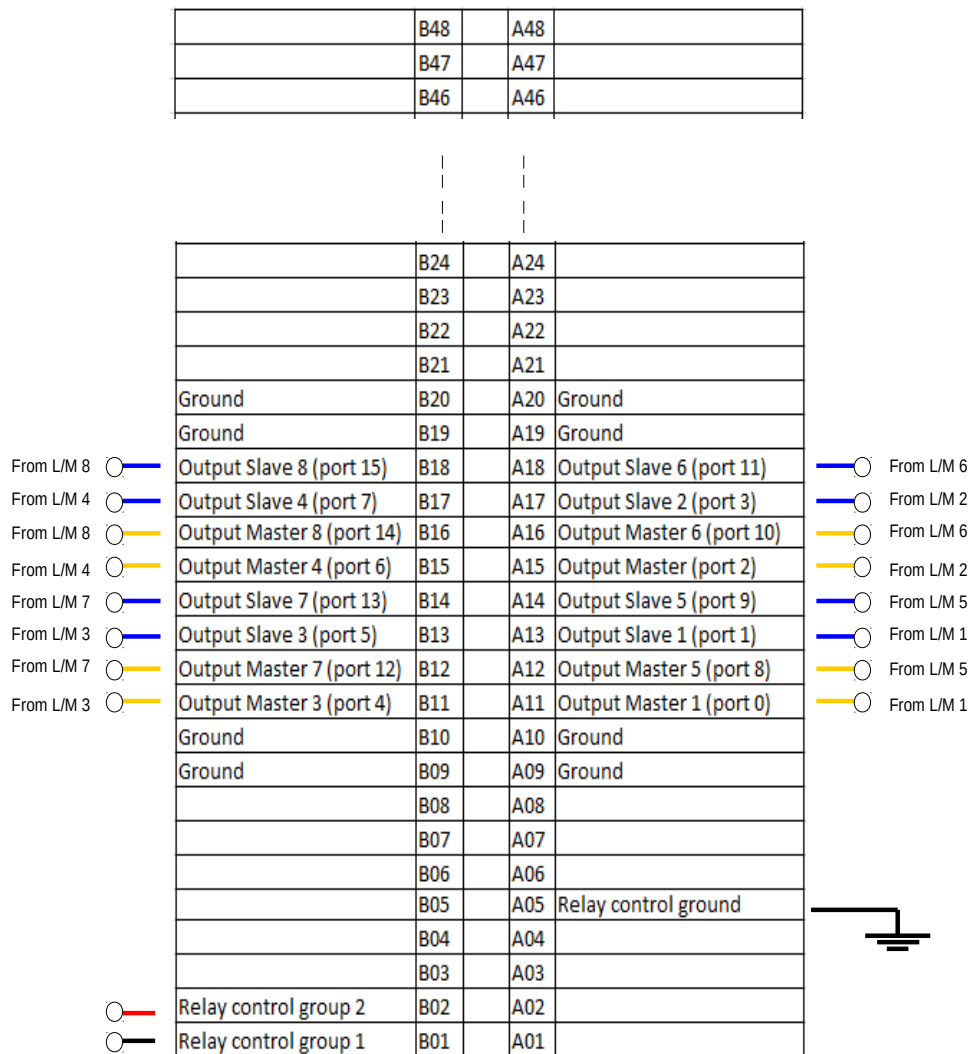


Figure 15: The ingoing distribution circuit. The figure shows only the mapping of the signals, since the Contec DTP-64 screw terminal is a premade device and further description is redundant. The port numbers in figure correspond to the ports that are addressed in the software.

4.4.4 Front and Back Panels

The front panel of the distribution box holds the DB-9 ports connecting with the L/M pairs. The connectors, for each L/M pair, are D-Sub connectors with 9 poles [4]. The mechanical sketch for the D-Subs is enclosed in Appendix A. The cable connecting the panel DB-9s with the L/M pairs is a standard DB-9 cable [5]. The latter mechanical sketch was lacking the dimensions of the port connector.

In order to cut the holes in the front panel, we had to use the mechanical workshop called “Mekanisk verkstad och industriell produktion” located in the A building. The distance between the DB-9 ports was 5 mm and the use of a computer numerical control (CNC) drill machine [6] was necessary to achieve such a level of drilling precision. In order for the CNC to cut the holes, a mechanical sketch of the front panel had to be created in a Computer Aided Three-dimensional Interactive Application



(CATIA) [7]. The CATIA software is able to output a mechanical sketch that the CNC's technician can use. The mechanical sketch used for the CNC can be found in Appendix A.

In the back panel, a DB-37 connector [8] is used to connect to the D/A card in the computer. The mechanical sketch for the D-Sub is the same as the one for DB-9. The mechanical sketch created for the back panel can be found in Appendix A. The mechanical sketch does not include the holes for the switches and the power lines. The latter holes were drilled using a hand operated drilling machine.

5 LIMITATIONS AND PROBLEMS

This section describes limitations and some of the problems that have been discovered and taken into consideration in the development of the product. All limitations and problems specified in this section have been confirmed when using the specific hardware and software specified in this document. Some of the limitations can quite effortlessly be taken care of, whilst other require either more effort or perhaps a replacement of components in either the hardware or software, or even both. No recommendations regarding how to solve the problems are given in this section but some ideas can be found in the Afterstudy [9].

5.1 A/D- and D/A Converters

The limitations found regarding the A/D- and the D/A converters are described in Sections 5.1.1, 5.1.2, and 5.1.3.

5.1.1 Sampling Clock

The limitation of the sampling clock is probably the most important since it has the largest impact on the accuracy that can be achieved when estimating the phase shifts enforced by the channel.

The maximum sampling frequency that can be used for the A/D and D/A converter is 100 kHz. This might not seem like a critical limitation, however, the problem occurs when using more than one channel, this is because the 100 kHz is divided with the number of channels that are used. For example, when 16 channels are active, that is all eight L/M-pairs are in use, one can in theory only achieve a maximum sampling frequency of 6250 Hz. In reality the sampling frequency can not be set arbitrarily to any value but is fixed to certain values, the maximum sampling frequency when using 16 channels is therefore 6024 Hz. The behavior described is valid for the A/D converter, whilst the D/A converter has a somewhat more undetermined behavior.

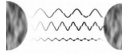
5.1.2 Transmitting Data

There is a limitation on the amount of data that can be put to the D/A converter, this limitation will not decrease the performance of the MIMO effect, although it will contribute to a rather annoying audible phenomenon that can be heard as a clicking noise for signals with a long time duration.

When data shall be transmitted from the D/A converter, there is a limit on the amount of data that can be put at once to the D/A engine. In theory, it should be possible to queue the data that exceeds this amount so that the signal still is sent continuously. Although, this has not been achieved and therefore when transmitting data that exceeds this maximum amount, one can hear some clicking noise. The limit of the amount of samples that can be put to the engine without the need for queuing is approximately 250 000, which translates to about 2.5 seconds, when using the standard setup, i.e. a sampling frequency of 6024 Hz and 16 channels.

5.1.3 Maximum Voltage Level of the A/D Converter

The maximum voltage level range of the A/D converter is $[-10, +10]$ V. This limitation has not been a large problem for this product, however if more L/M-pairs of the same kind are to be used in a future product, these limits are quite low. With only eight L/M-pairs it is perfectly possible to achieve a signal strength in the point of focus that exceeds this range. It is of great importance that the sound level from each L/M-unit is high enough to estimate the channel but at the same time be low enough to not exceed the limits when the powerful MIMO effect is being used. As a side note one shall be aware of that the power supply used for the product delivers voltages in the interval $[-11, +11]$ V and that the



A/D converter probably can not tolerate these voltages for an unlimited amount of time without being damaged, therefore it is important to be careful and not use a too high sound level from the L/M-units in the MIMO array.

5.2 L/M-Units

The limitations found regarding the L/M-units are described in Sections 5.2.1 and 5.2.2.

5.2.1 Different Characteristics for Master and Slave Unit

There is a difference in the characteristics between the master and slave unit and if this is ignored, one will most probably experience a nondeterministic system behavior.

When using the L/M-units as loudspeakers, there is a rather large difference in the sound level that is produced from the master- and slave unit, when they are given the same signal. Because of this and since there also is an upper limit on the input voltage levels, this leads to that the slave units have a smaller operating distance range. Another characteristic that will contribute even more to this smaller range is the fact that the slave units, when operating in microphone mode, are also less perceptive. This means that they are more noise sensitive and will need a larger sound level than the master units in order to record data that is useful in the channel estimation. When using the system in the far field, this different characteristics of the master and slave units results in that the loudspeakers must produce quite high sound levels in order for the system to work appropriately, ear protection is therefore advised.

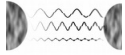
5.2.2 Different Characteristics between Different L/M-Pairs

There is a difference in the characteristics between different L/M-pairs. If this is not taken into consideration the system will probably not perform as intended.

The characteristics that differ and are of importance between the different L/M-pairs is basically the amplification. There is a difference in amplification in both the loudspeaker- and microphone mode. This leads to that the weakest L/M-pair will set the distance range boundary, between the MIMO array and the point of focus. Therefore one must make sure that the recorded data from the weakest L/M-unit is good enough for estimating the channel.

5.3 Software

The software has not really been a limitation by itself, although the software that is used, that is, the code written by the project group, Windows 7, drivers, MATLAB and the DAQ toolbox, might be the cause of some of the limitations and problems that was described in Section 5.1. For example, there have been some problems regarding the stability of the communication with the D/A and A/D converter, this might have influenced to the data transmission limitation, described in Section 5.1.2.



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A DISTRIBUTION BOX SCHEMATICS

A.1 D-Subs9/37

DS - 9 FCN

D-SUB Solder Type 9,15,25,37,50
 Nickel or Tin or Gold Plated
 Color(Black,Blue,White or Others)
 Female

ELECTRICAL CHARACTERISTICS:
 CONTACT CURRENT RATING:1AMPERES.
 DIELECTRIC WITHSTANDING VOLTAGE:AC 500V R.M.S
 INSULATION RESISTANCE:500 MEGOHMS MINIMUM AT DC 500V.
 CONTACT RESISTANCE:30 MILLIOHMS MAXIMUM.
 OPERATING TEMPERATURE:-40°C TO +70°C.

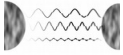
MATERIALS:
 CONTACTS:BRASS.
 INSULATOR:P.B.T.U.L 94V-0 RATED.
 SHELL:COLD ROLLED STEEL.

FINISHES:
 CONTACTS:GOLD FLASH .
 SHELL: NICKELPLATED OR TIN PLATED
 OR ZINC PLATED OR GOLD PLATED OR OTHERS.
 PLASTIC: BLUE OR BLACK OR WHITE OR OTHERS.

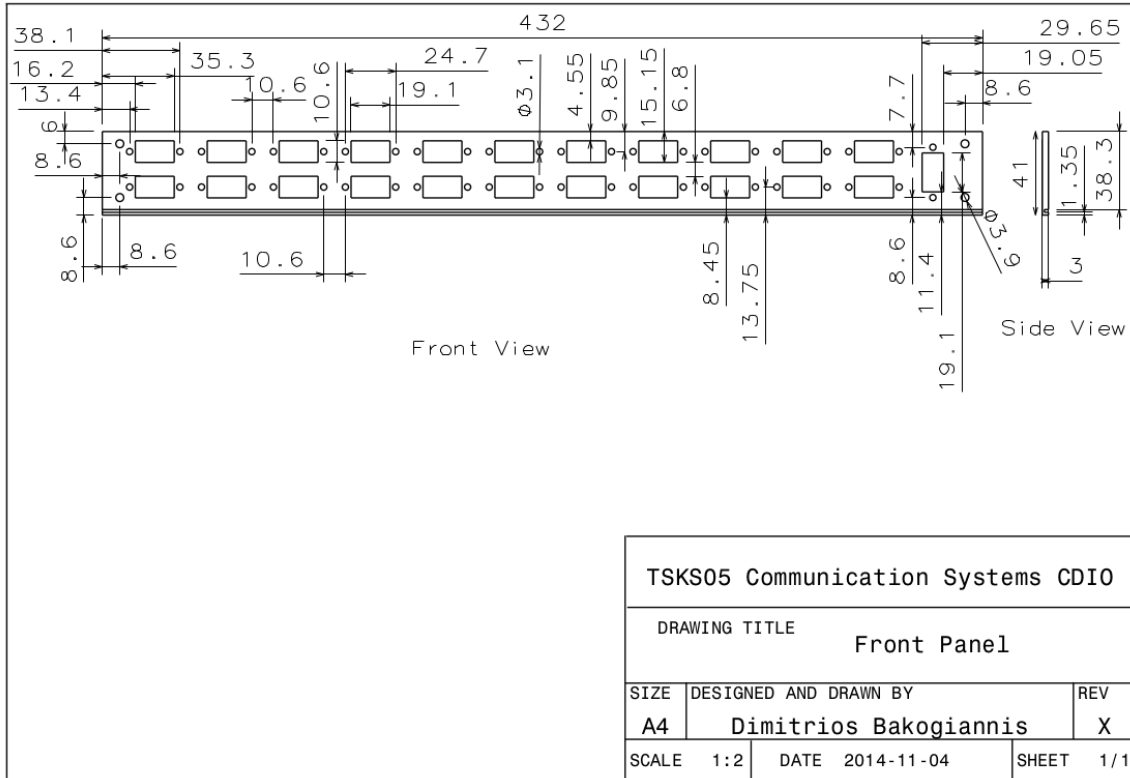
POSITION	A±0.3	B±0.2	C±0.2	D±0.2	E±0.2	F±0.2	G±0.2	H±0.1
DS-9Pin	30.85	25.00	16.40	7.90	14.95	12.55	2.77	2.84
DS-15Pin	39.25	33.35	24.70	7.90	14.95	12.55	2.77	2.84
DS-19Pin	44.80	38.86	30.00	7.90	14.95	12.55	2.77	2.84
DS-23Pin	50.30	44.30	35.30	7.90	14.95	12.55	2.77	2.84
DS-25Pin	53.10	47.04	38.52	7.90	14.95	12.55	2.77	2.84
DS-37Pin	69.40	63.50	54.85	7.90	14.95	12.55	2.77	2.84
DS-50Pin	67.08	61.05	52.25	10.90	14.95	15.37	2.77	2.84

RoHS Compliant

TOLERANCES OTHERWISE STATED	REVISIONS			CUSTOMER P/N:
	REV	DESCRIPTION	DATE	
X ±0.3				CH P/N.:
XX ±0.2				
XXX ±0.1				TITLE: D-SUB 9pin FEMALE CONNECTOR
XXX ±0.05				DRAW BY: <i>REB</i>
ANGULAR ±2°				CHECKED:
				UNIT: mm
				SCALE:
				DATE: 2002.04.18



A.2 Front Panel





A.3 Back Panel

