

# RoomPerfect DSP room correction

by

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Chief Technology Officer

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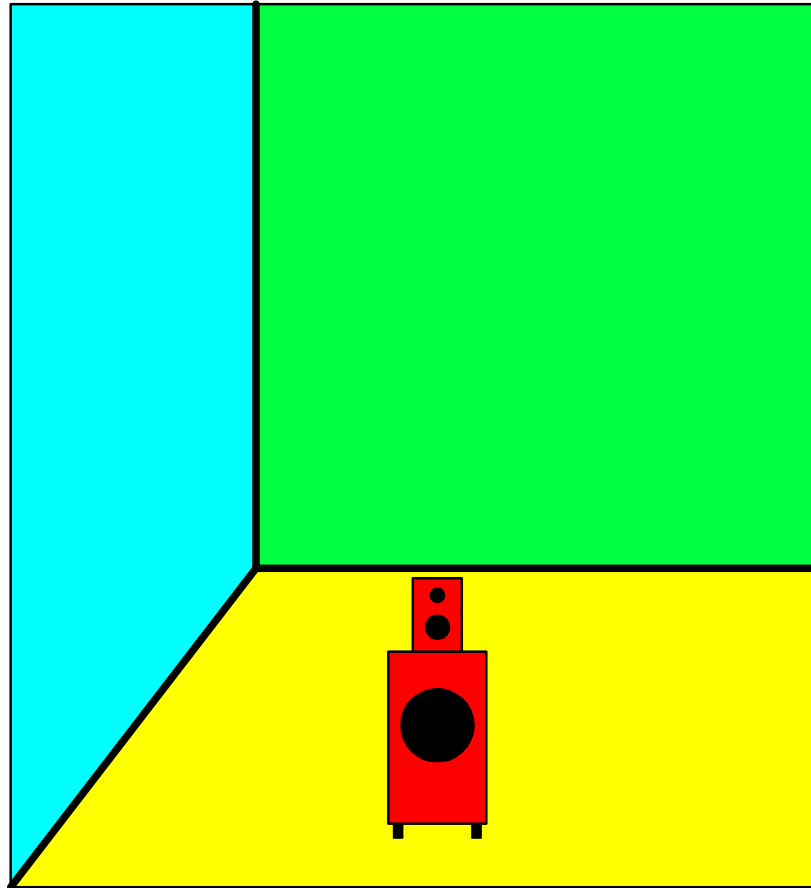
Lyngdorf Audio

Denmark

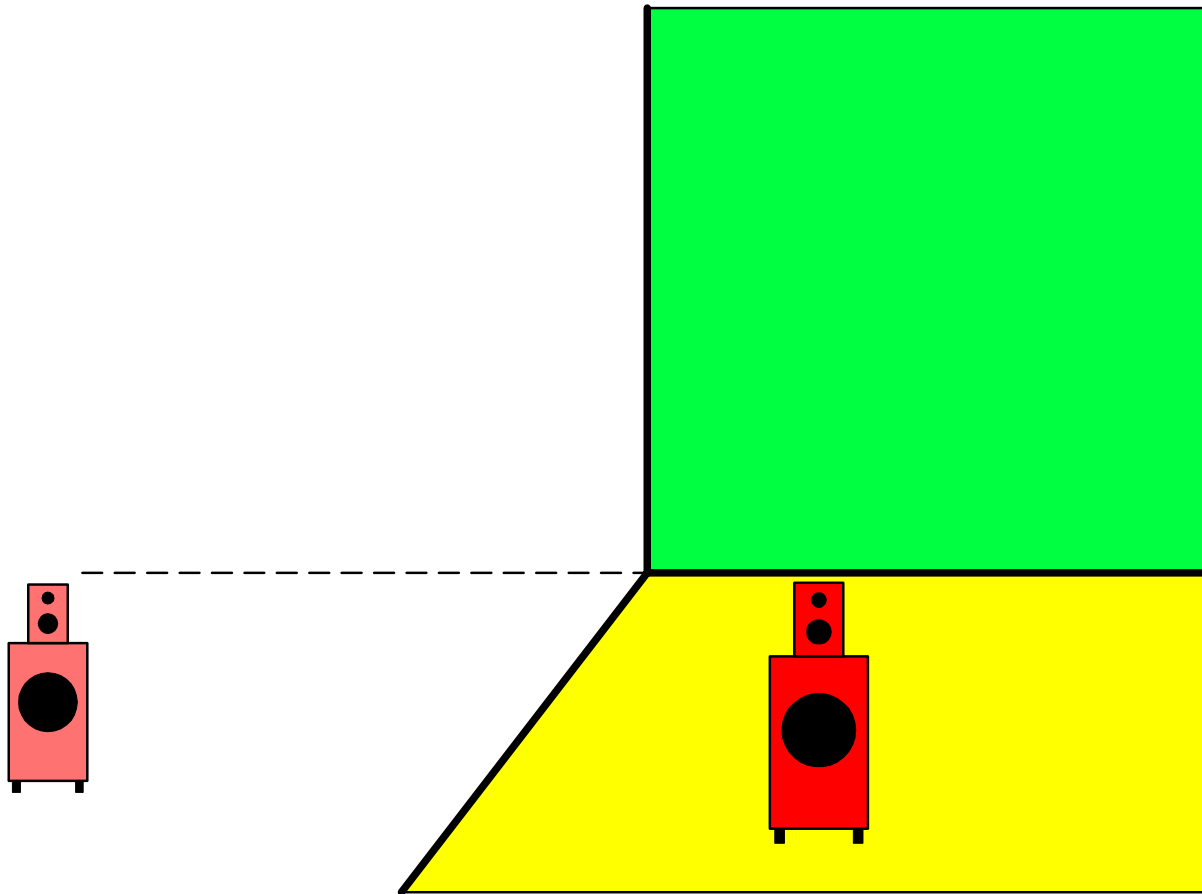
# Agenda

- Introduction
- Sound field in a room
- System principle
- Microphone positions
- Automatic target calculation
- Gain limits
- Correction filter for an actual listening room
- Conclusion

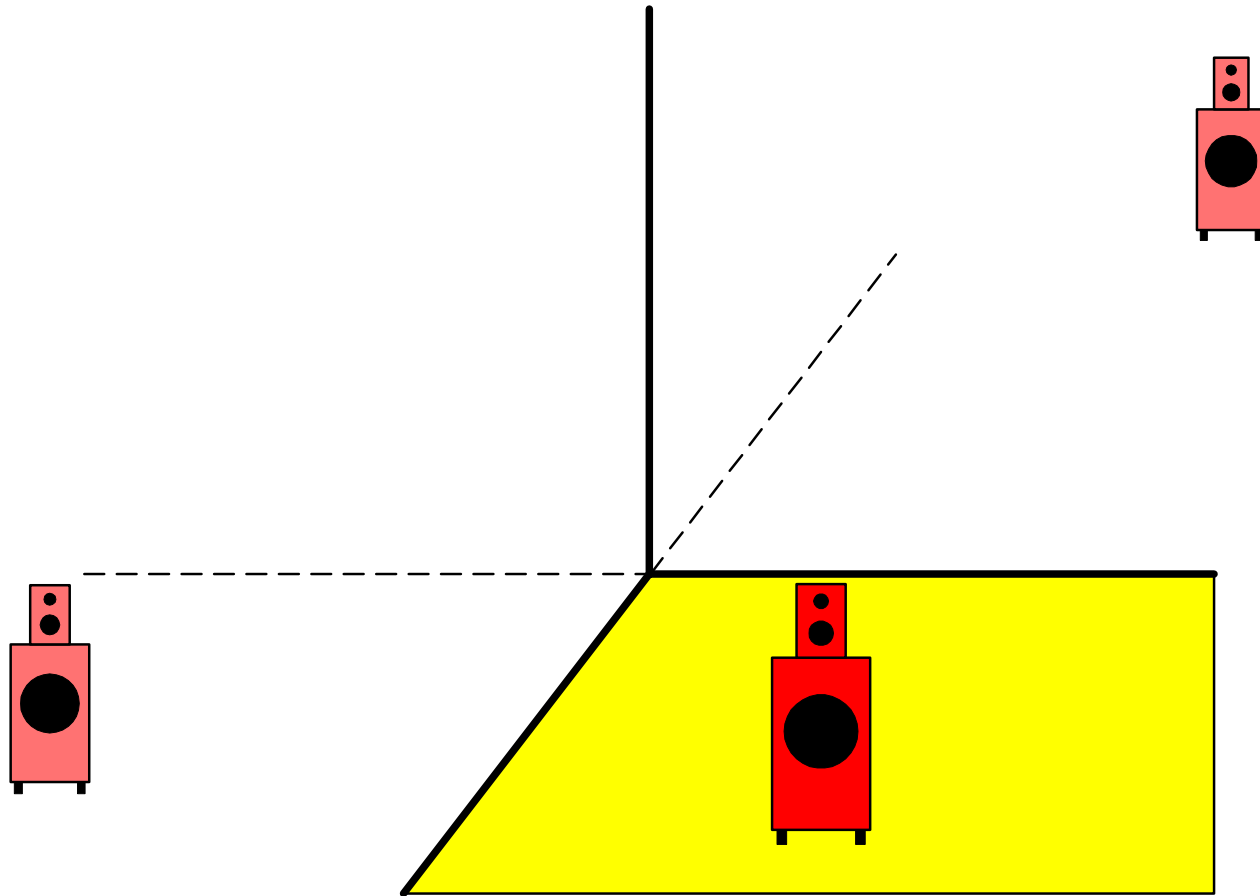
# What you see



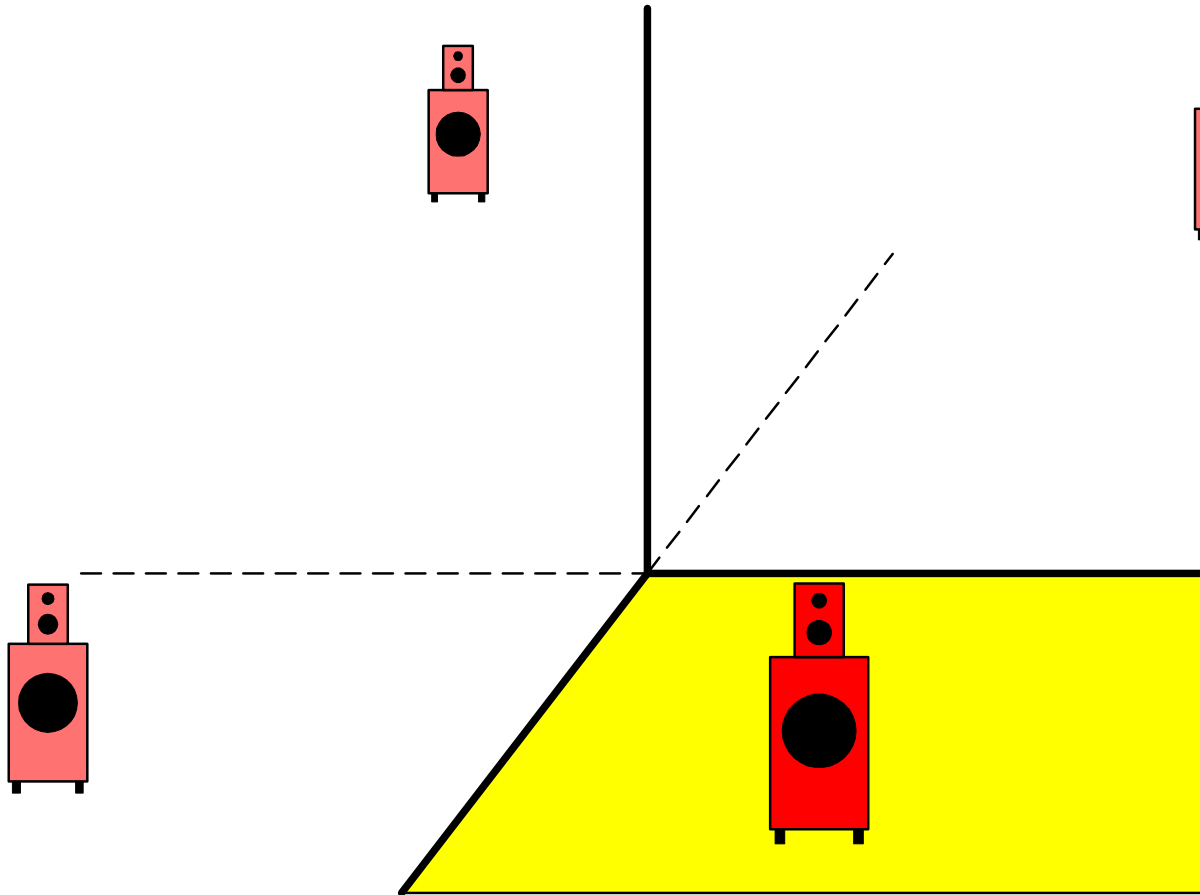
# What you hear!



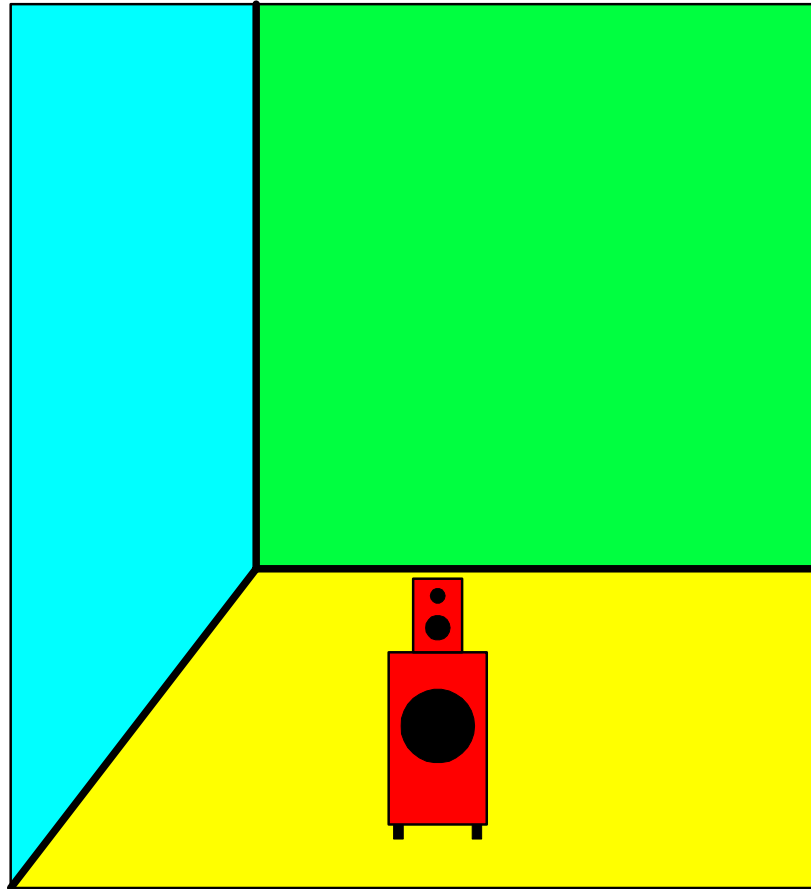
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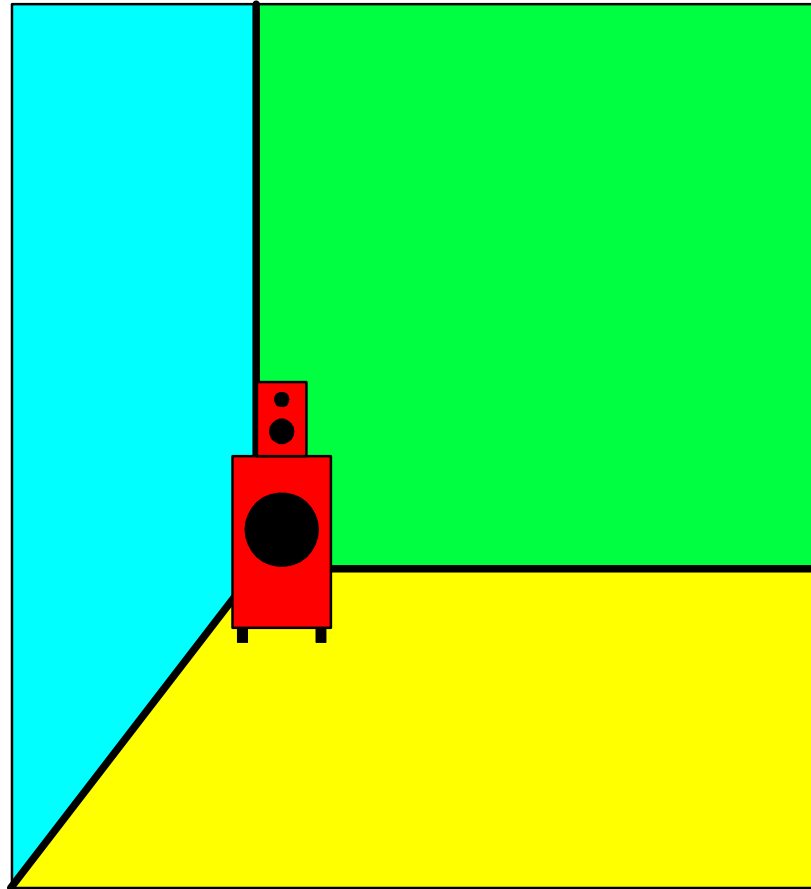
# What you hear!



# What you see!

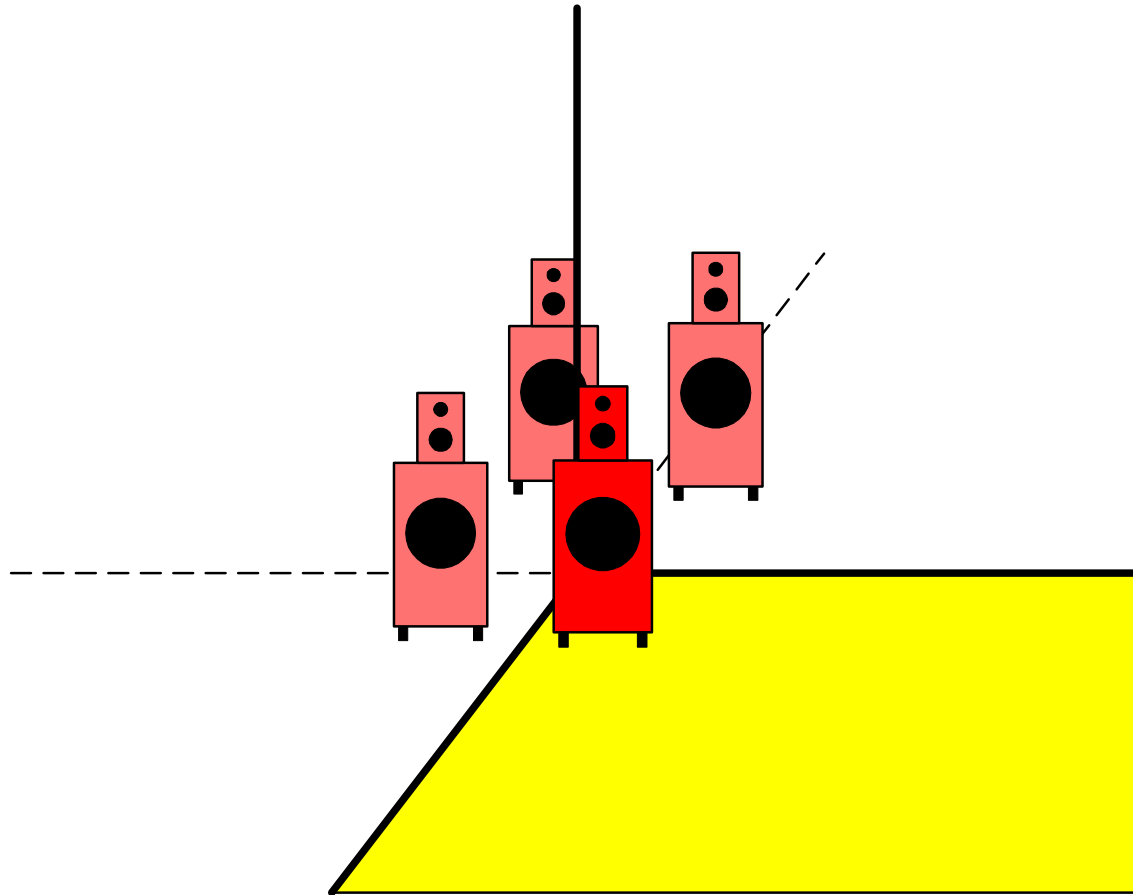


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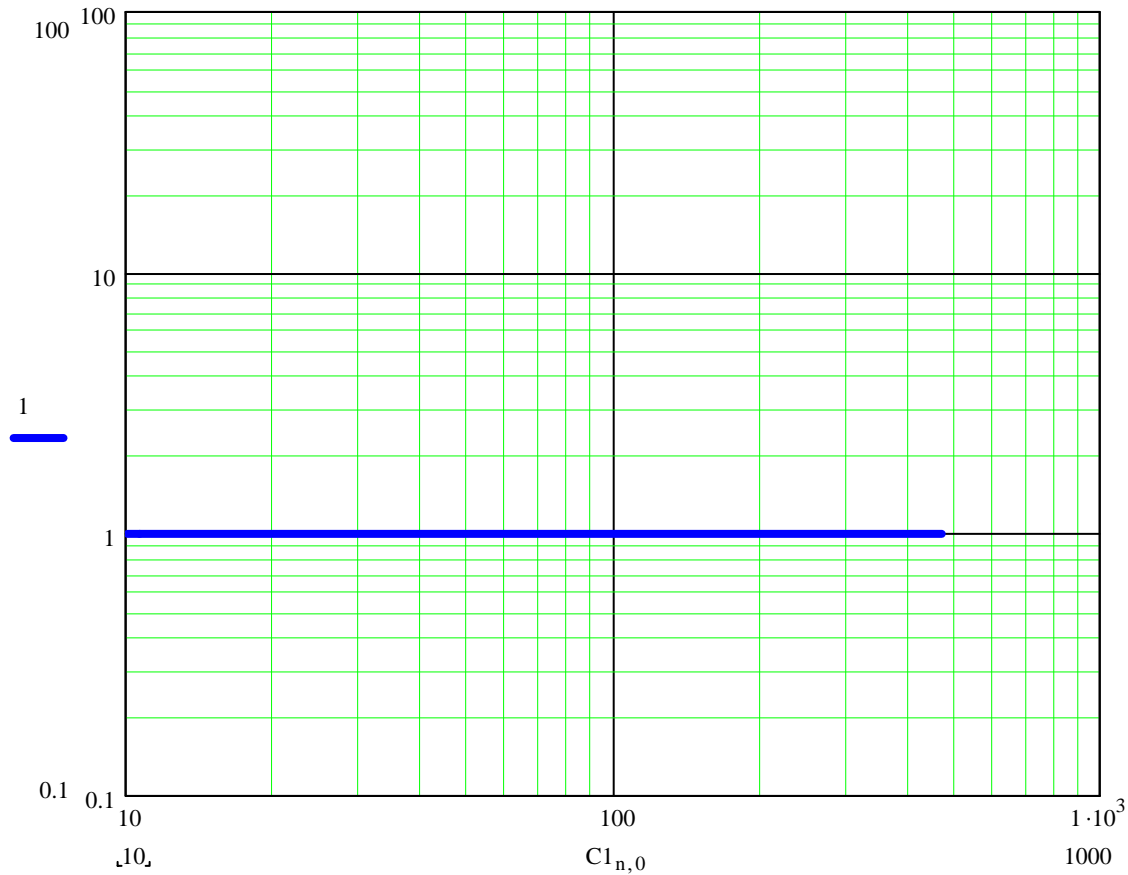




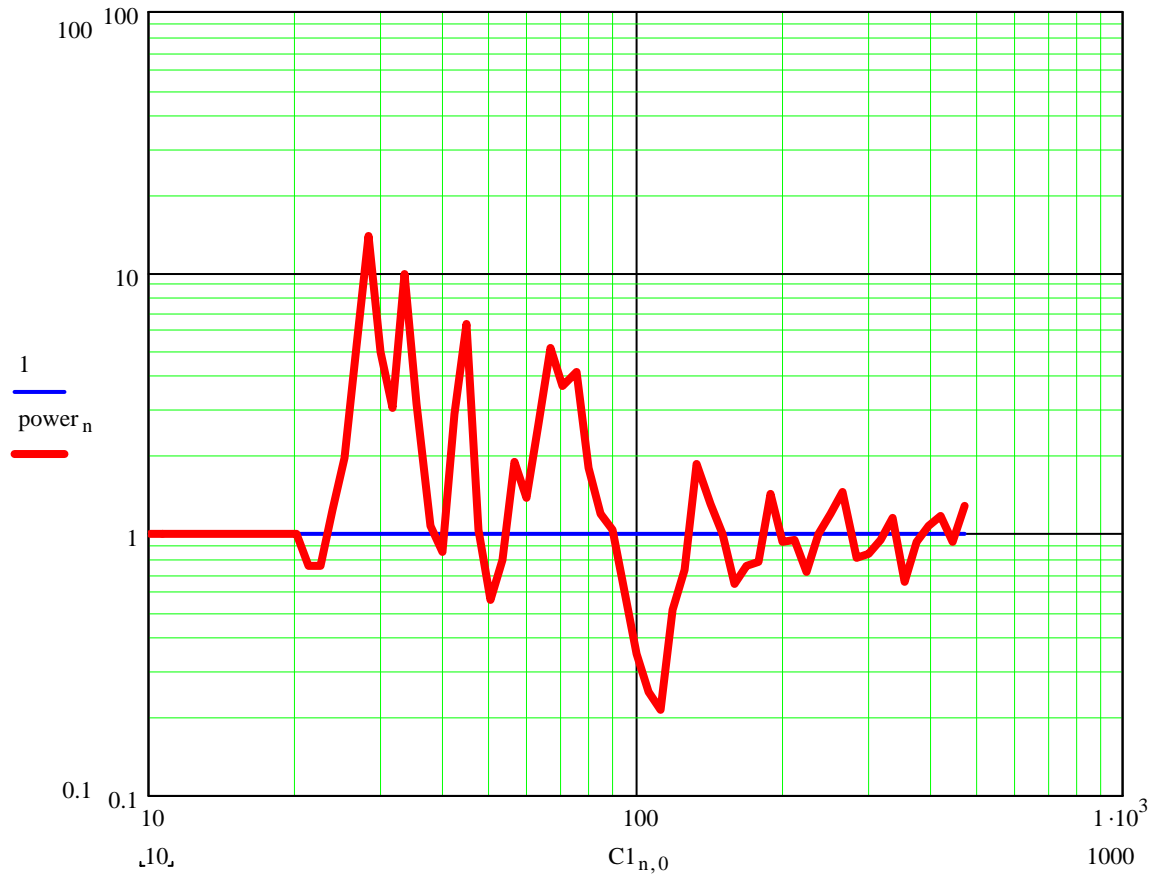
# What you hear!



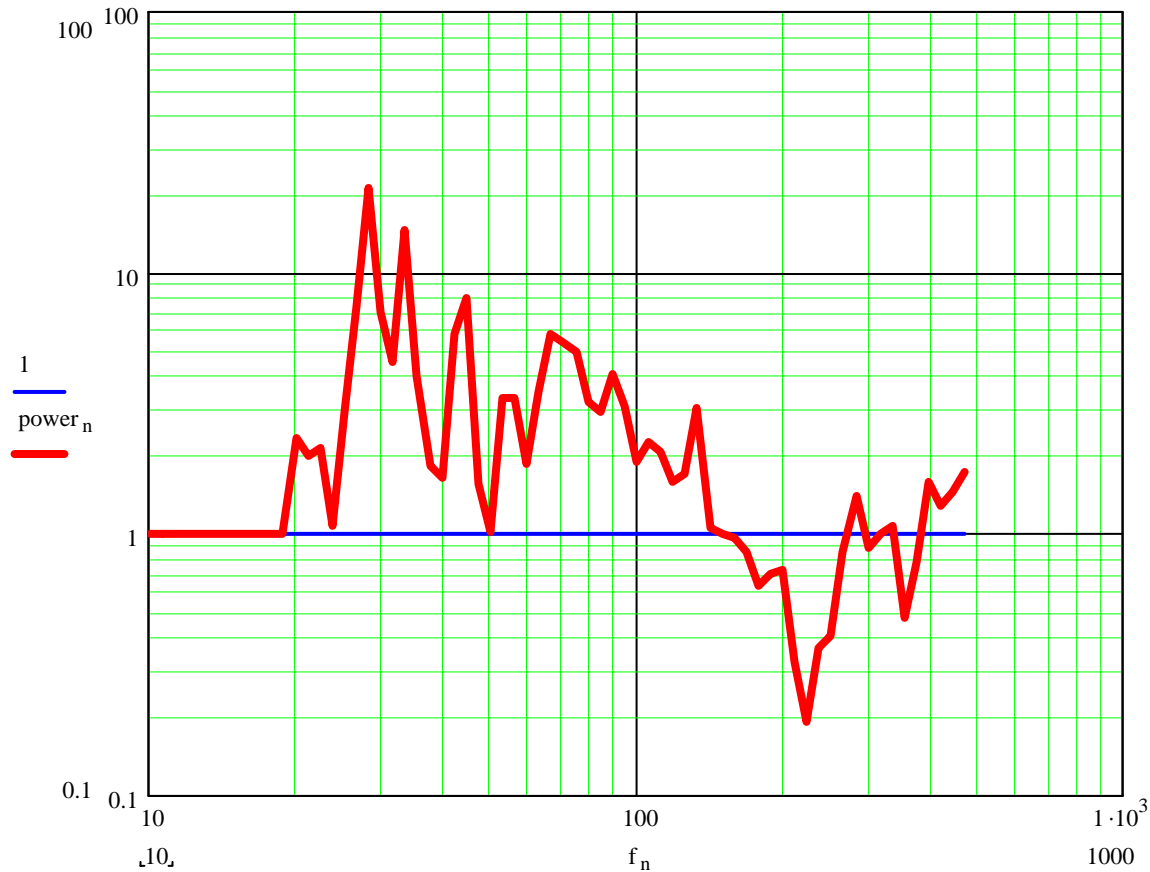
# Power Output [watt] – Free Field



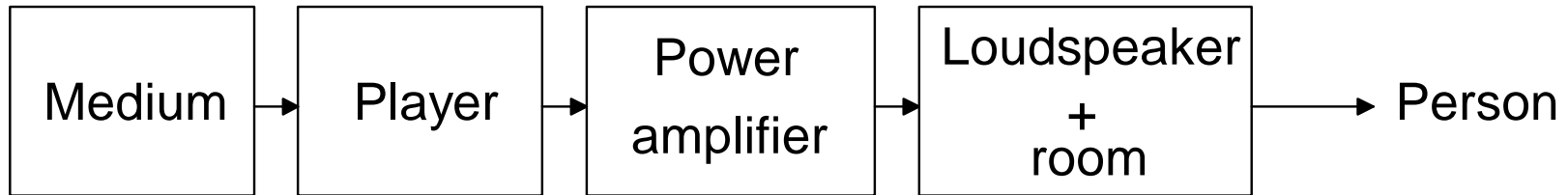
# Power Output [watt] – Ref. Pos.



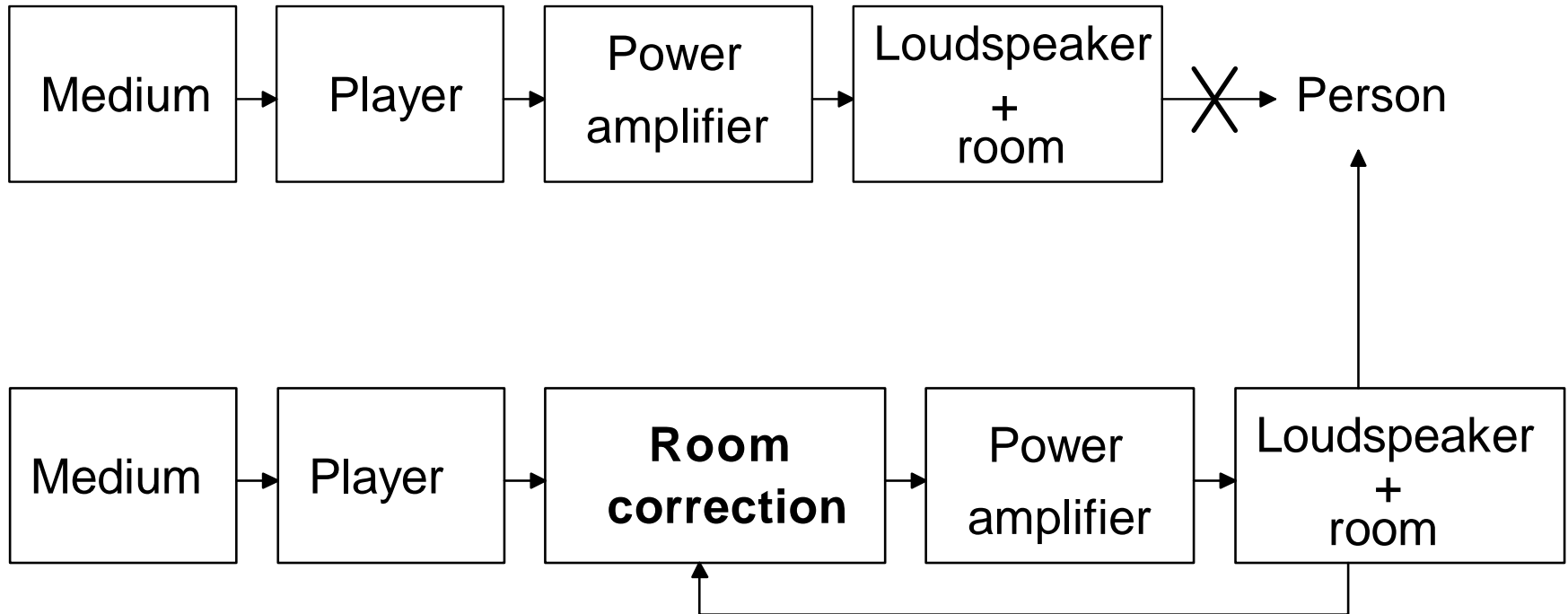
# Power Output [watt] – Corner



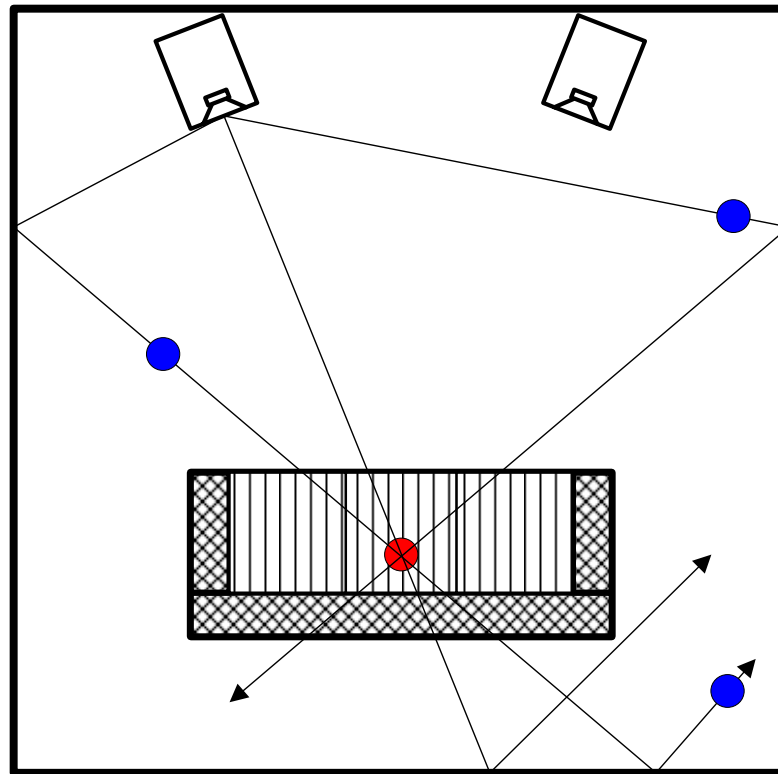
# Signal Path



# Signal Path

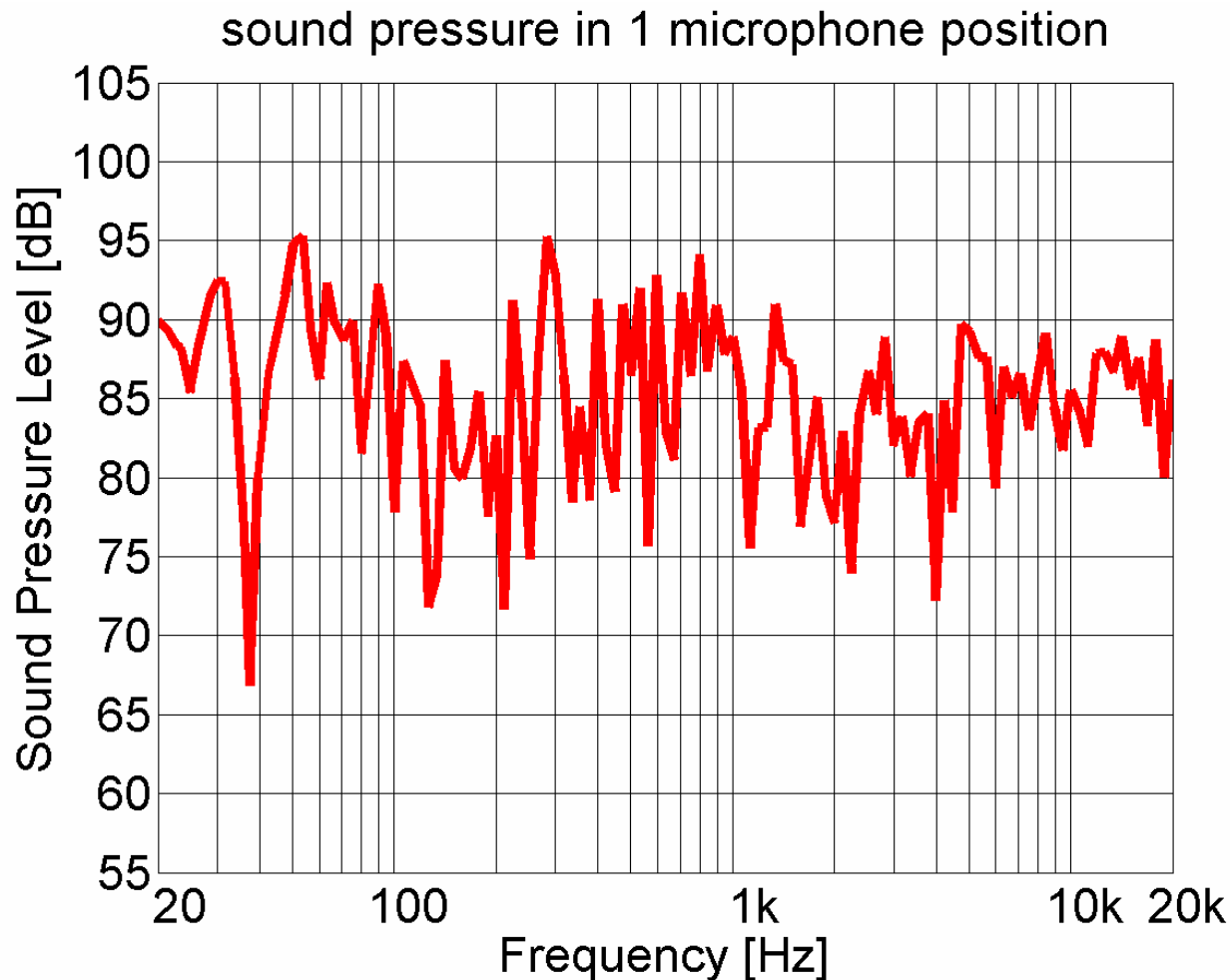


# Measured Sound Pressure



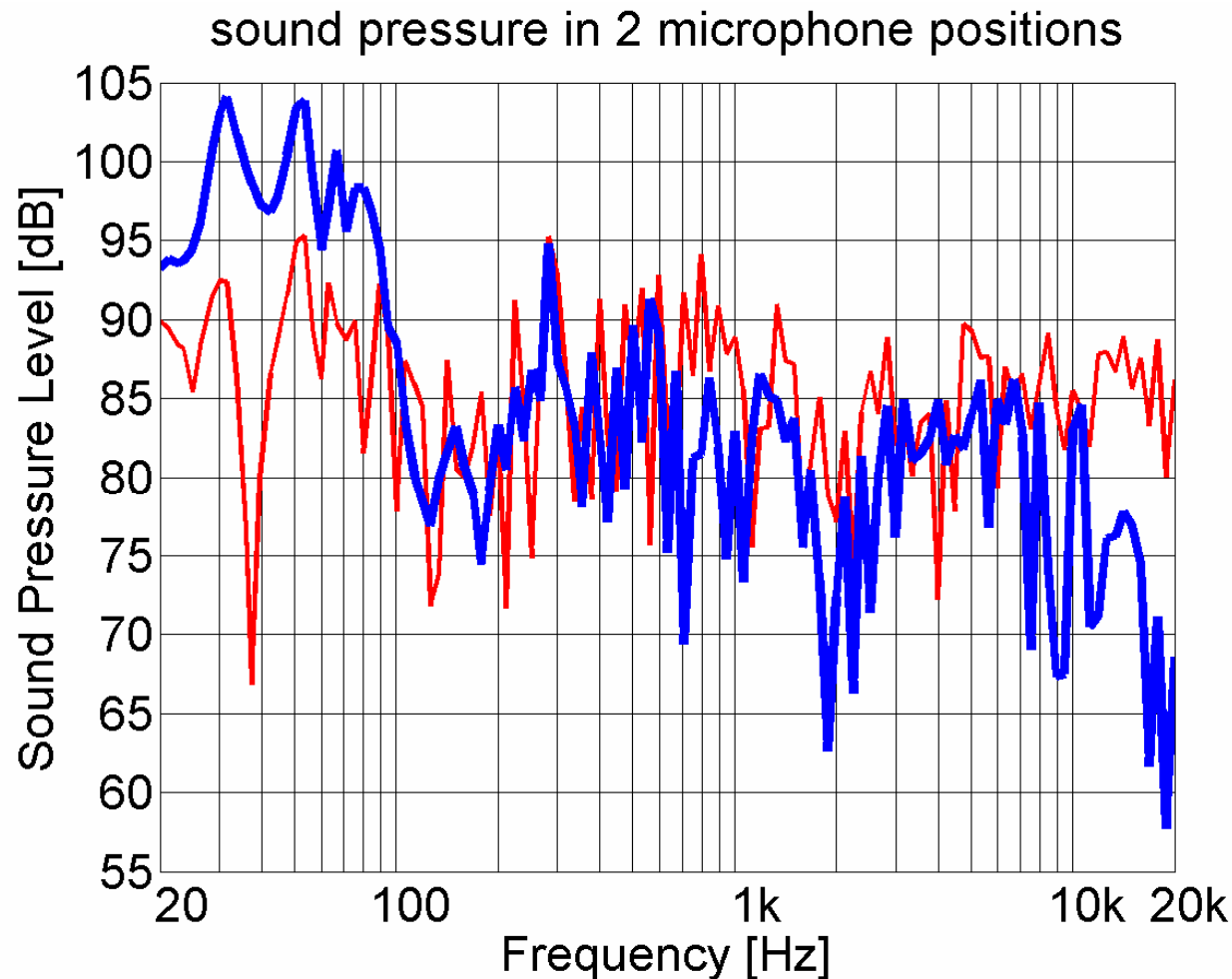
Red = Listening Position  
Blue = Room Positions

# Measured Sound Pressure

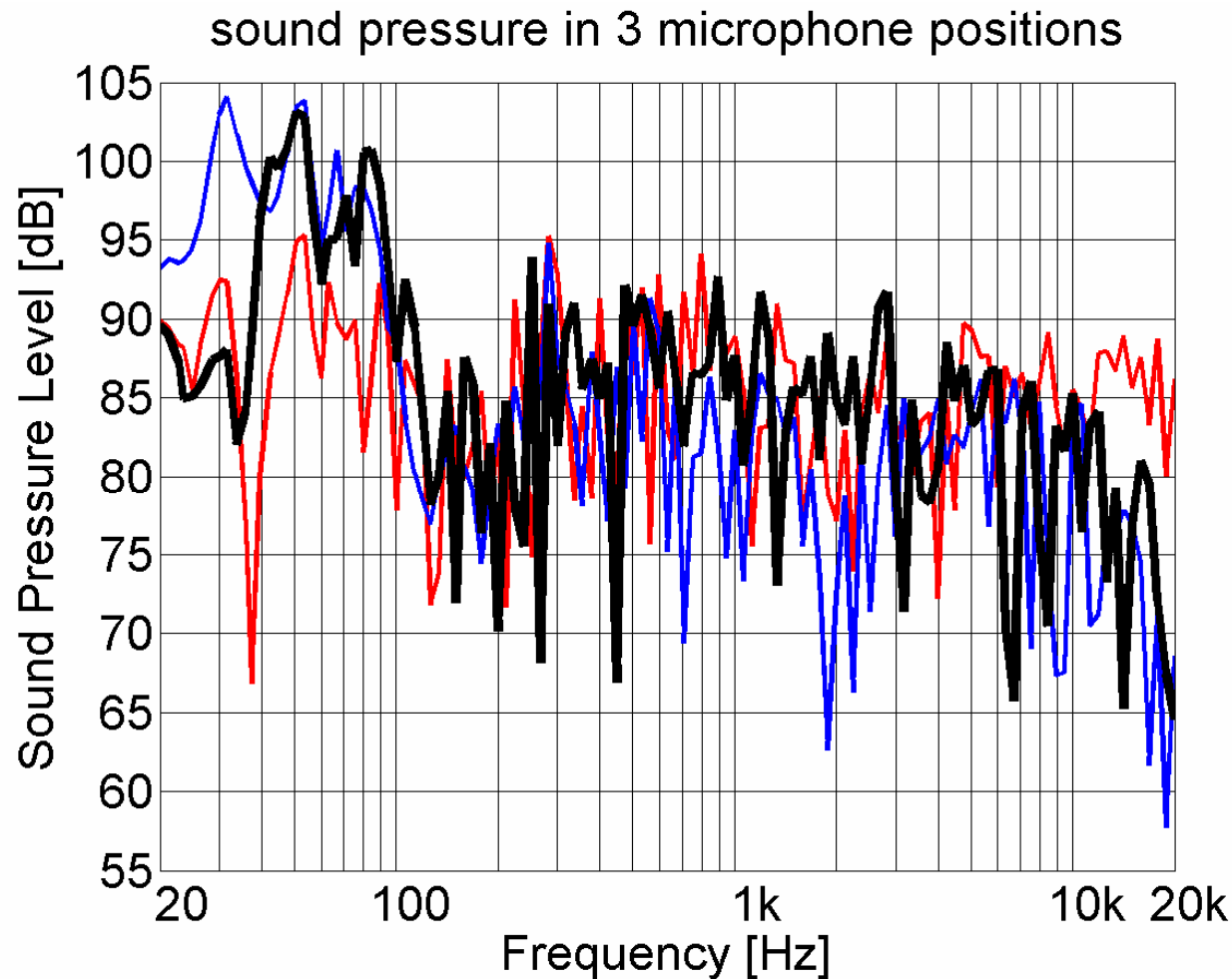




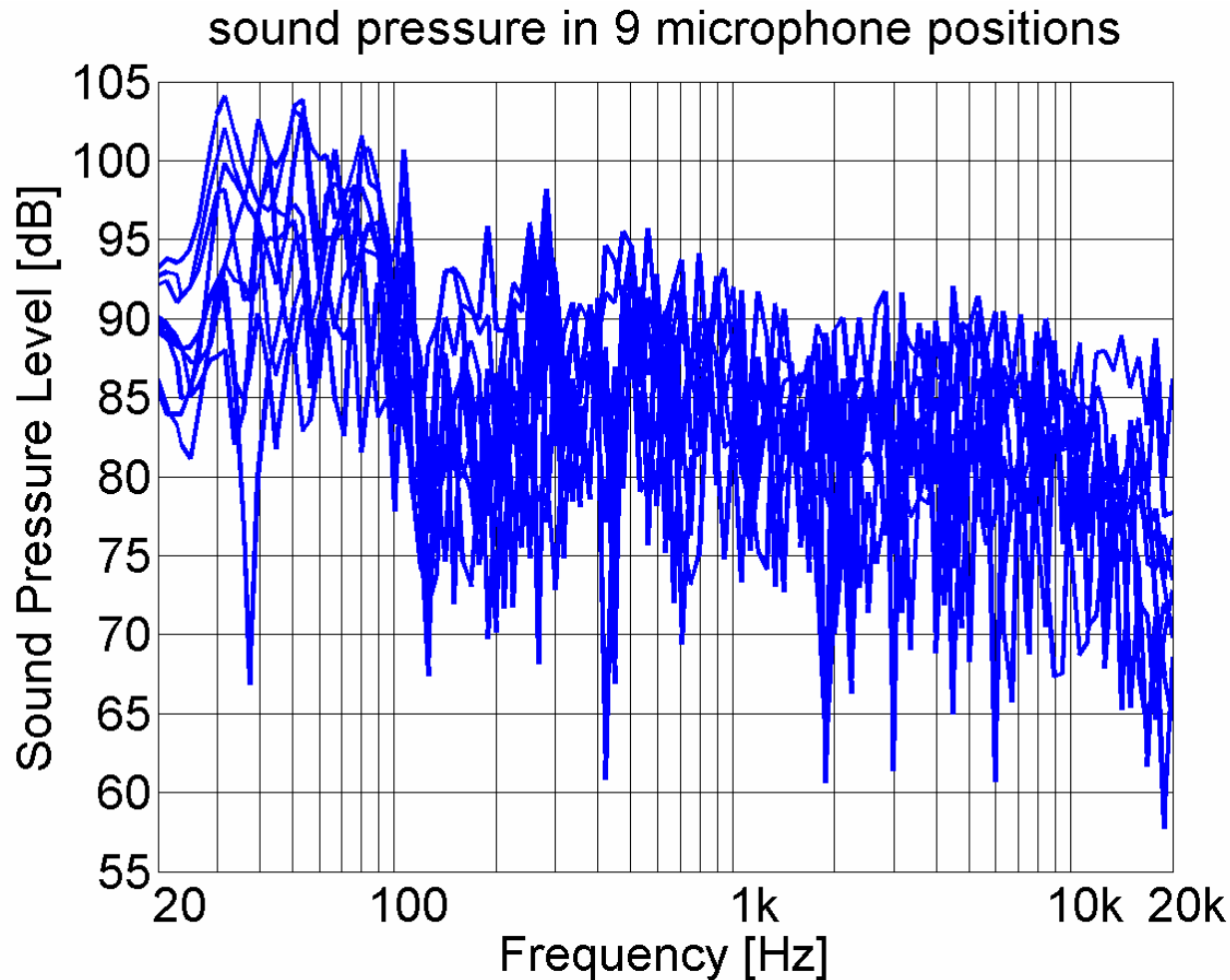
# Measured Sound Pressure



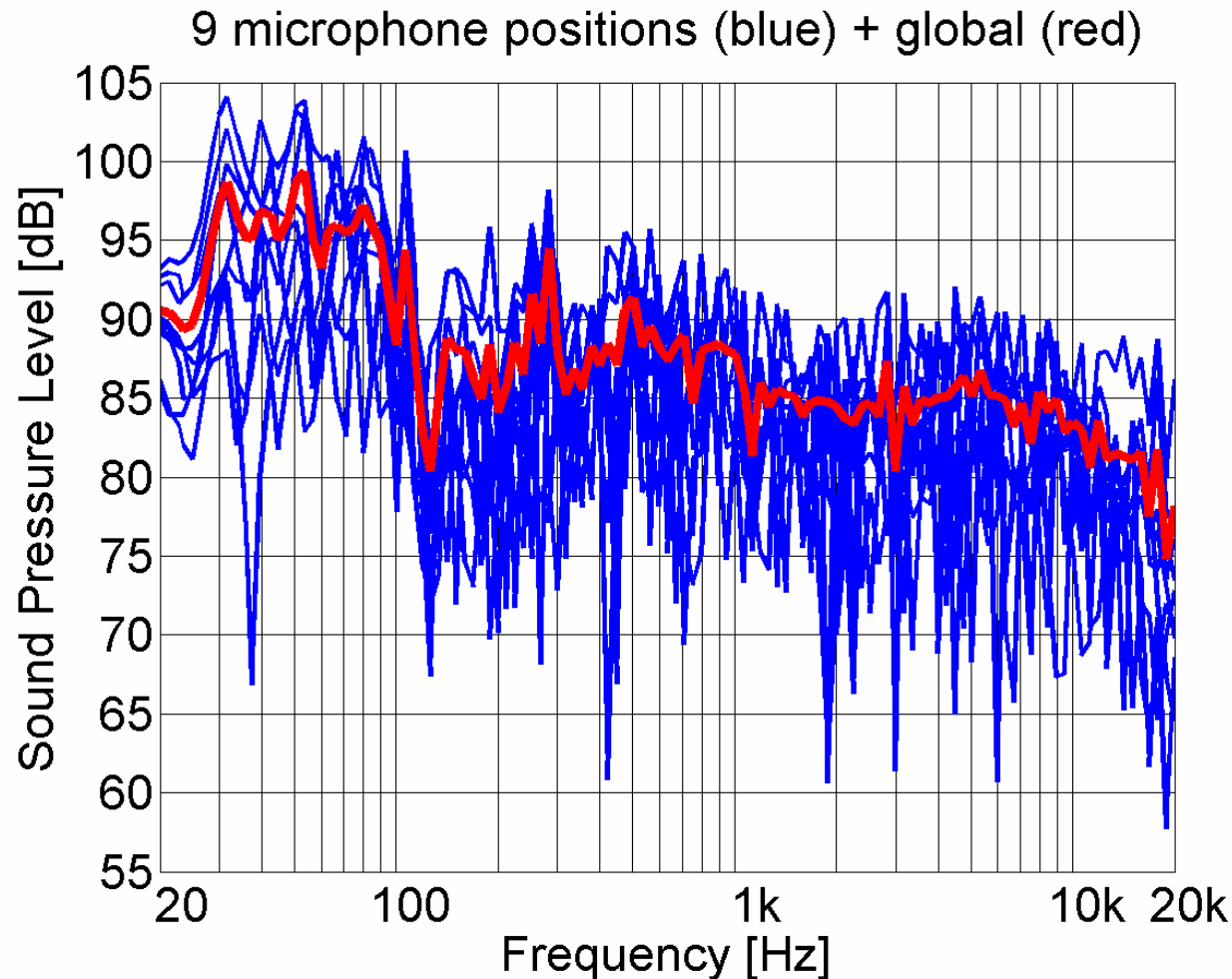
# Measured Sound Pressure



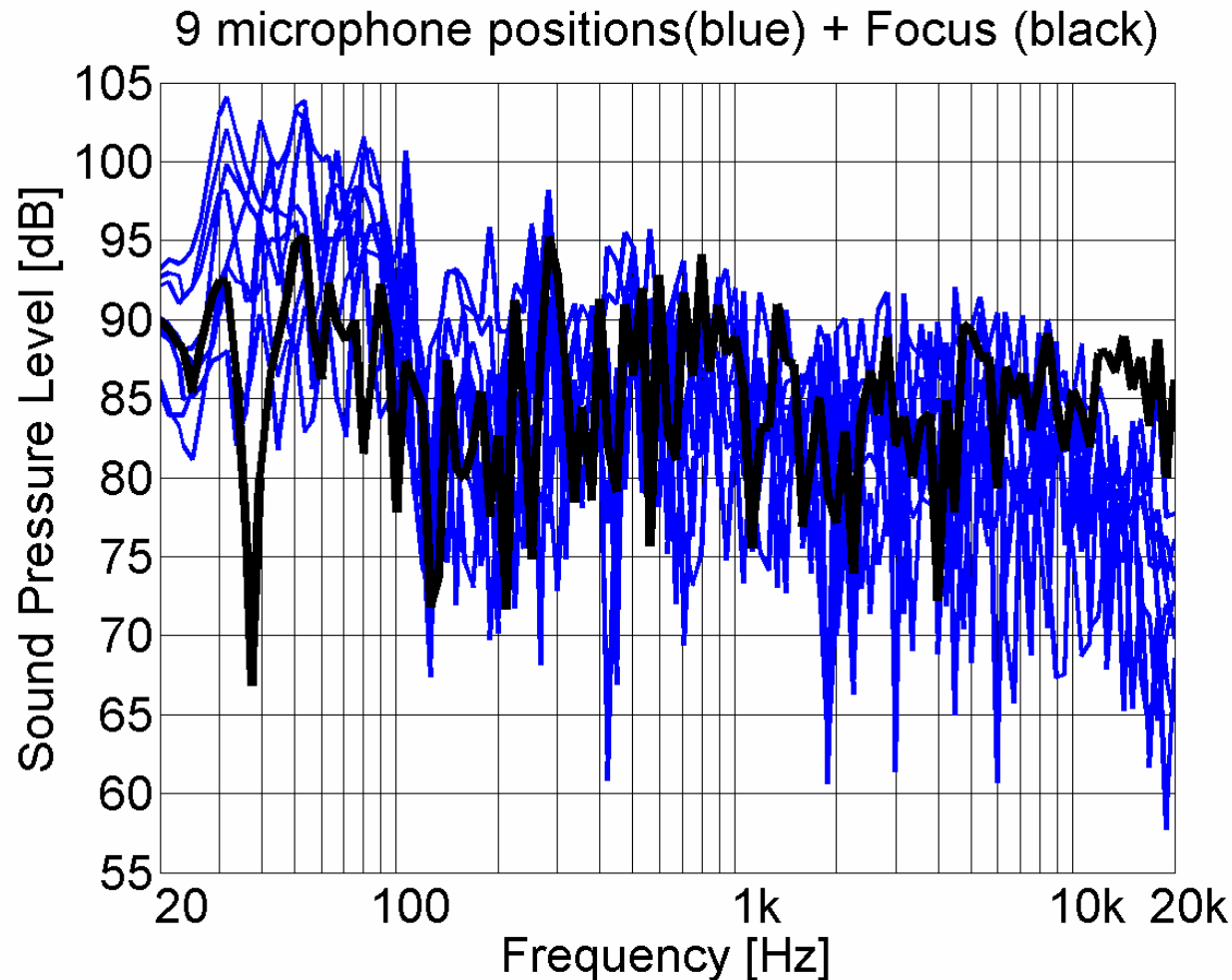
# Measured Sound Pressure



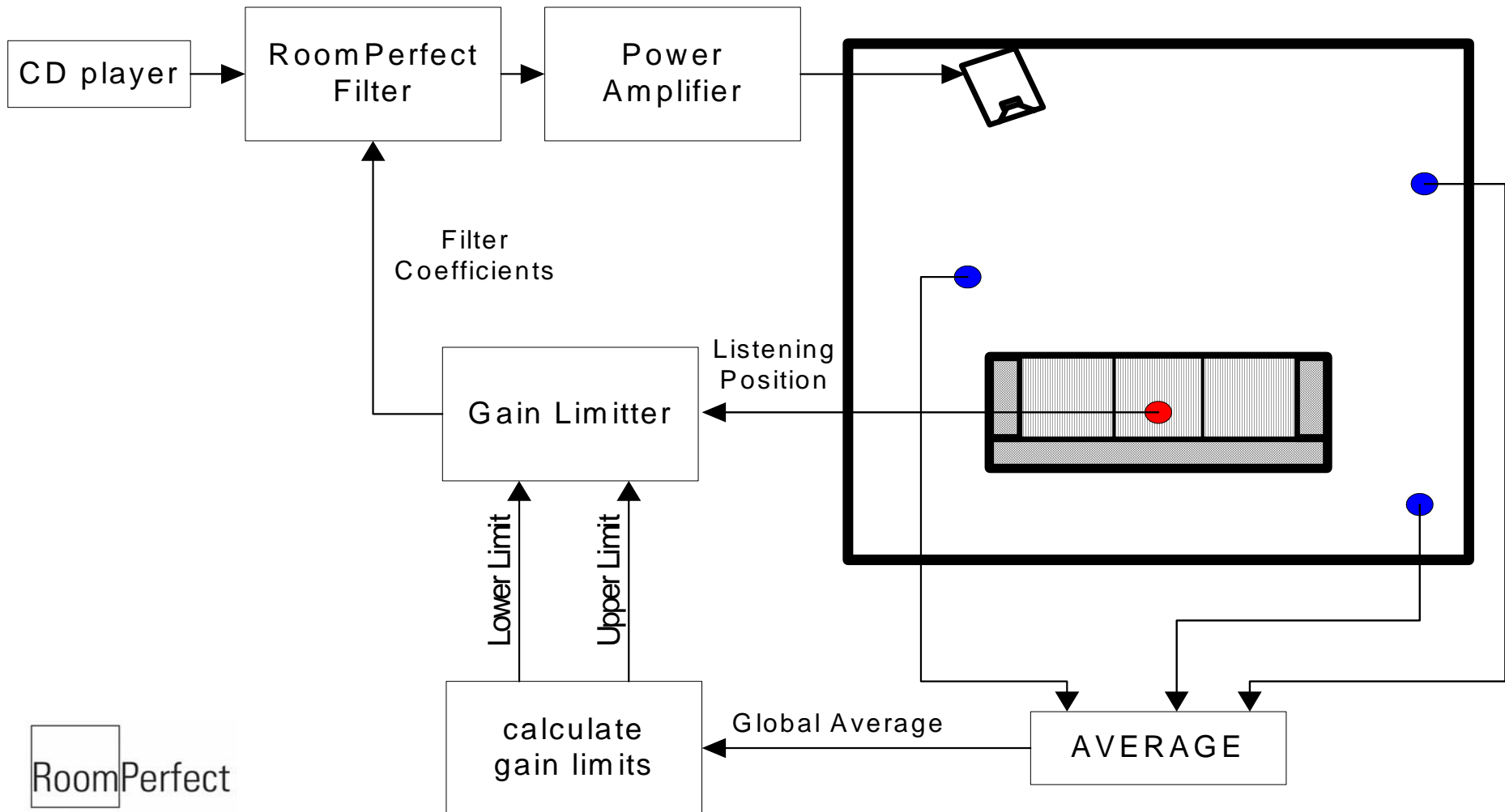
# Power averaged Sound Pressure



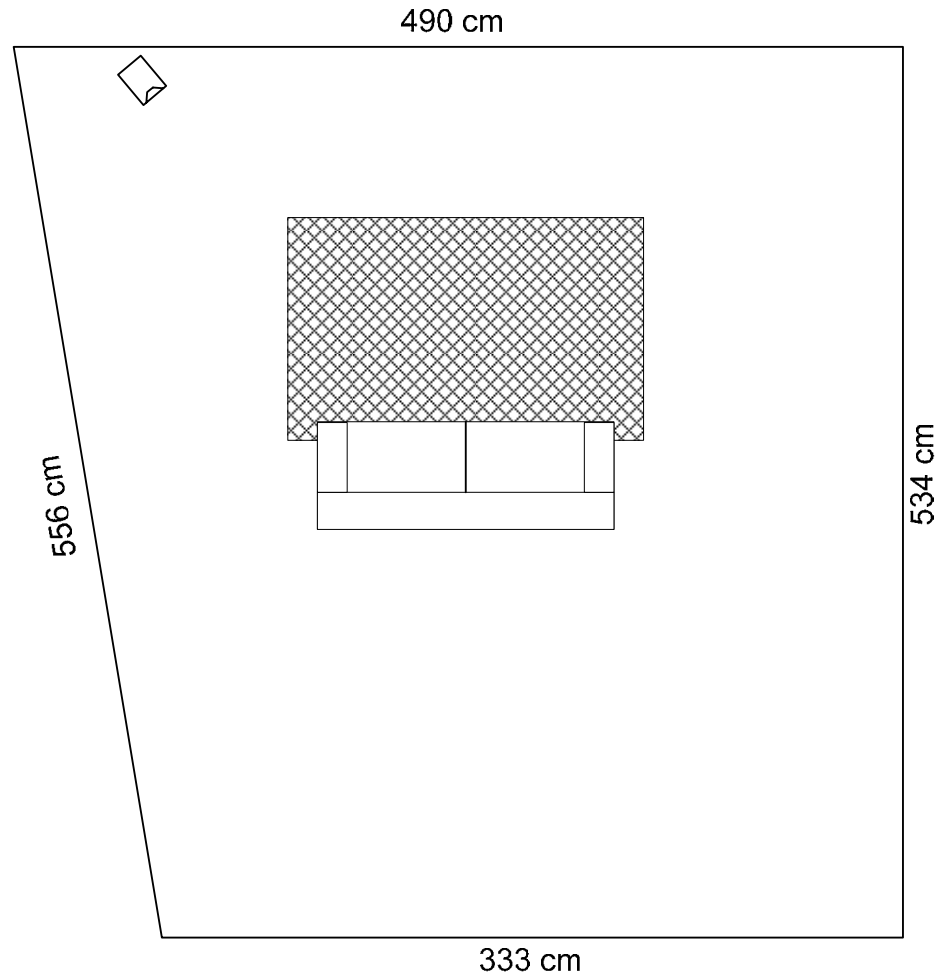
# Sound pressure at the listening position



# System principle



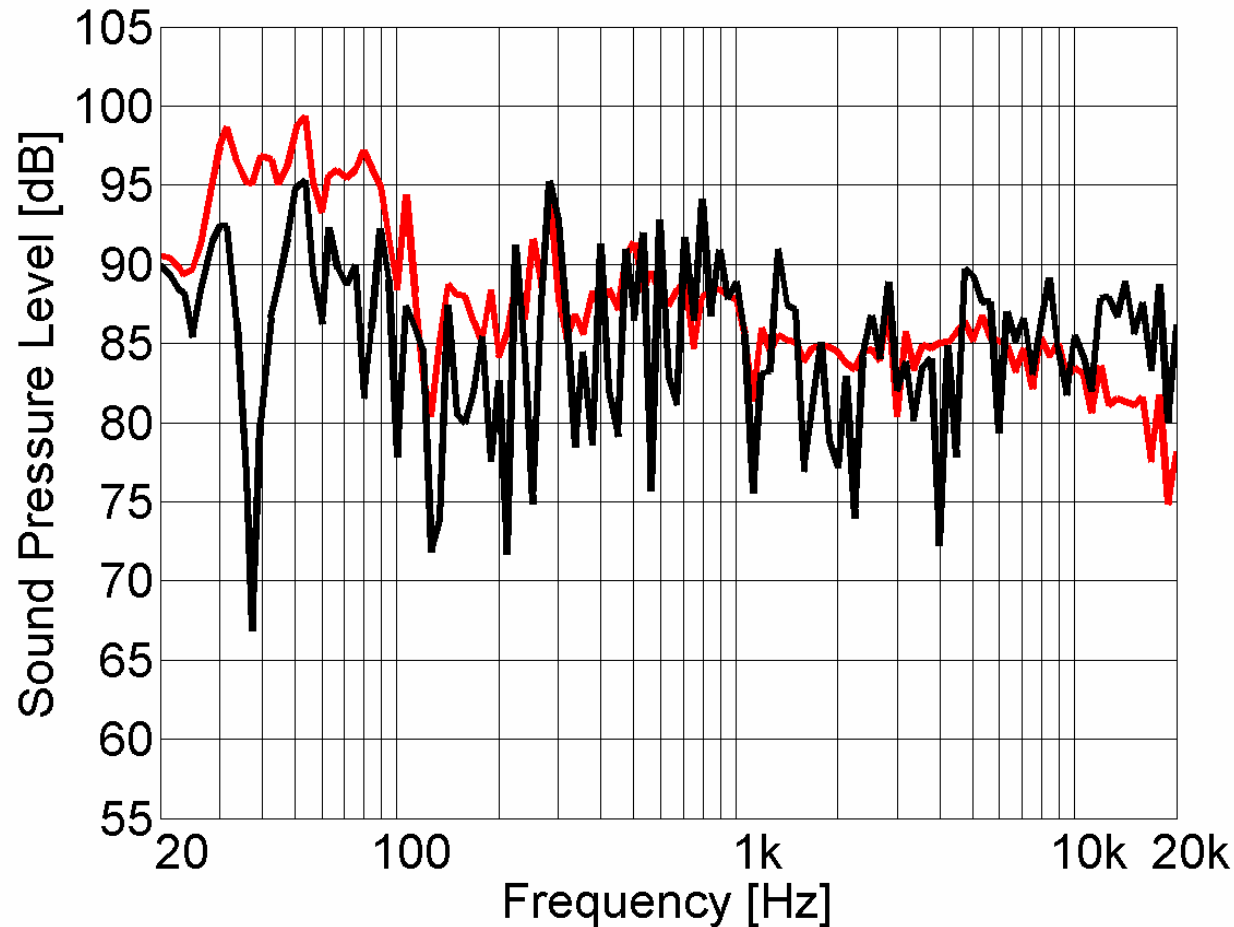
# Measuring in a listening room



# Microphone Positions

## 1 Position

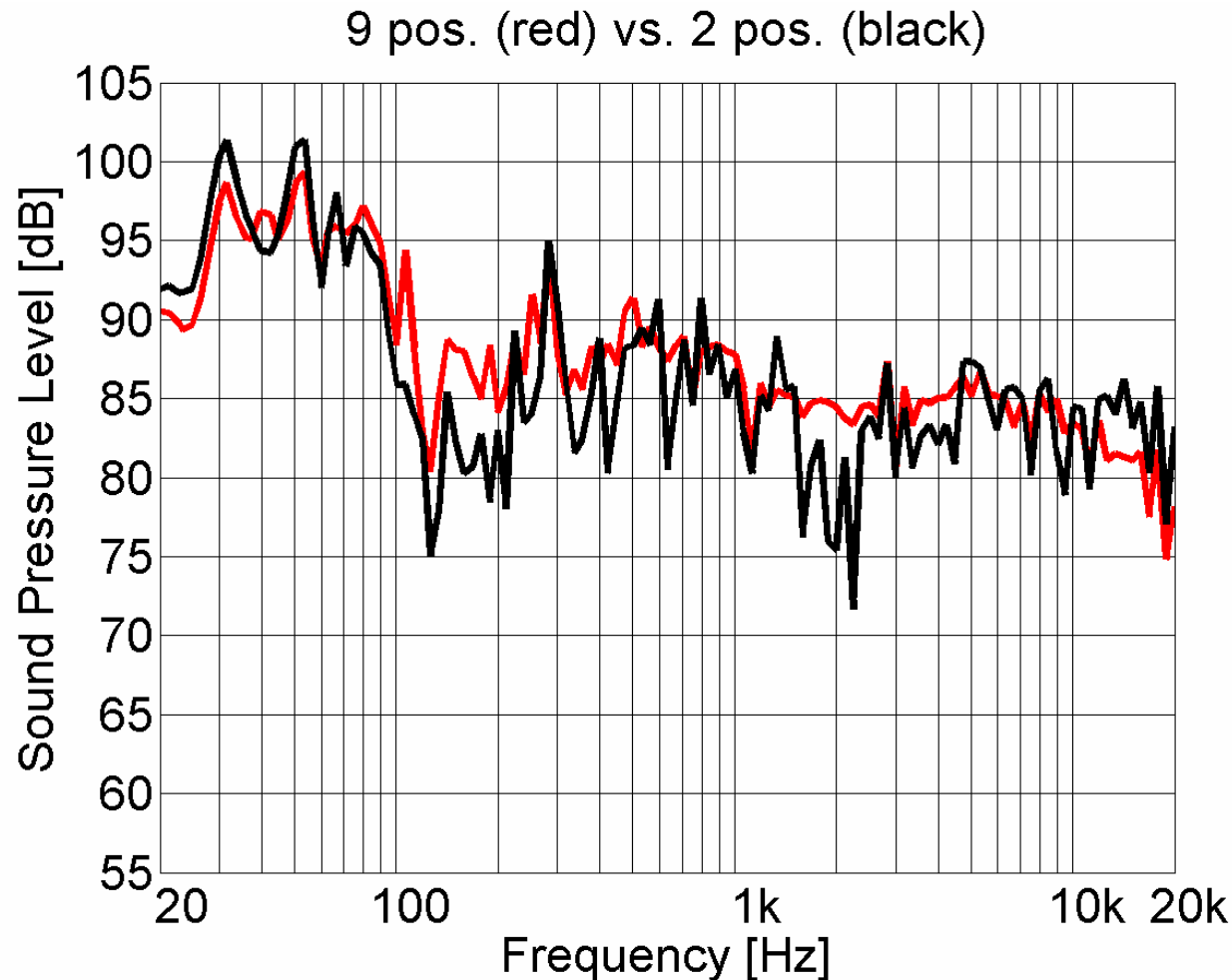
9 pos. (red) vs. 1 pos. (black)





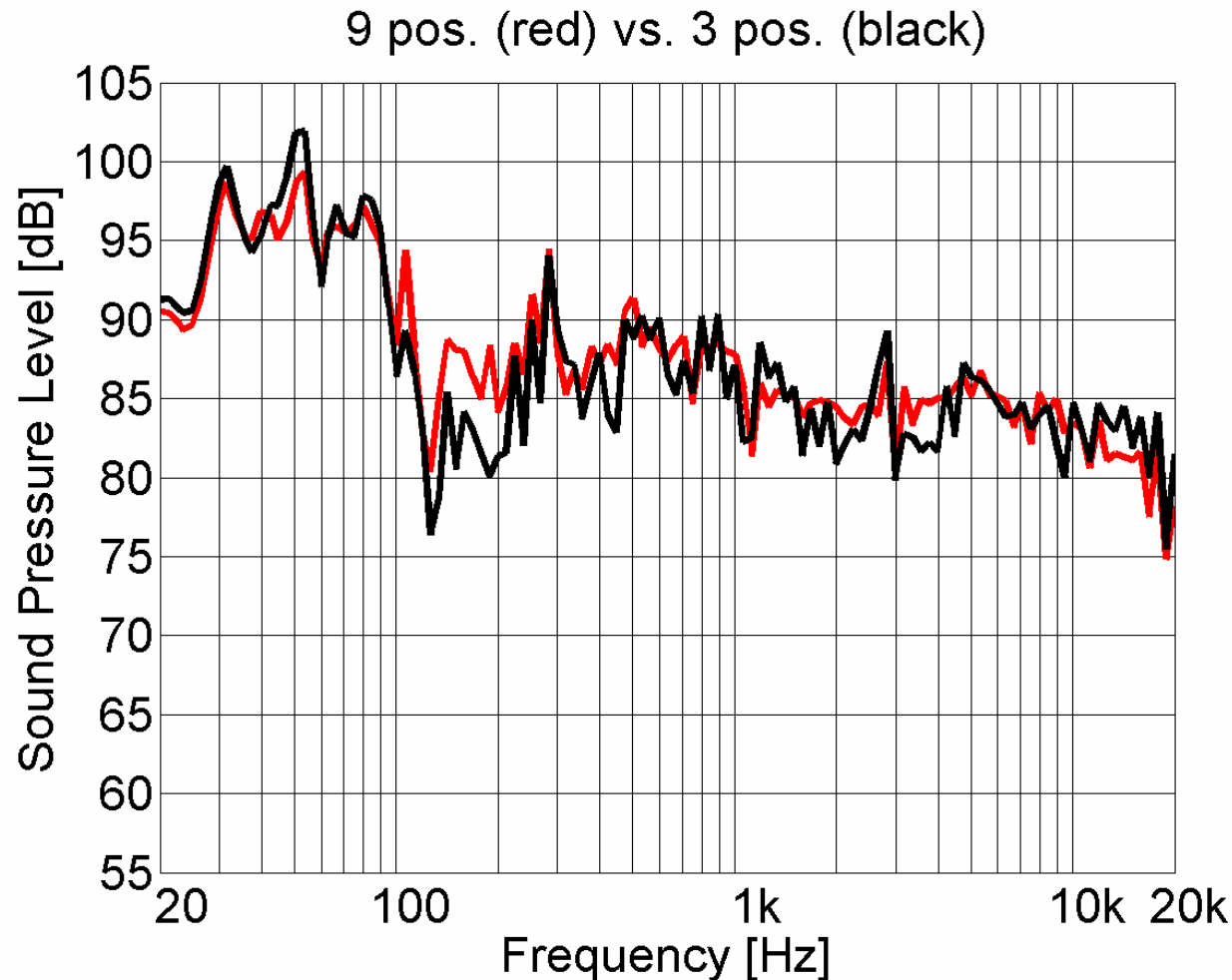
# Microphone Positions

## 2 Positions



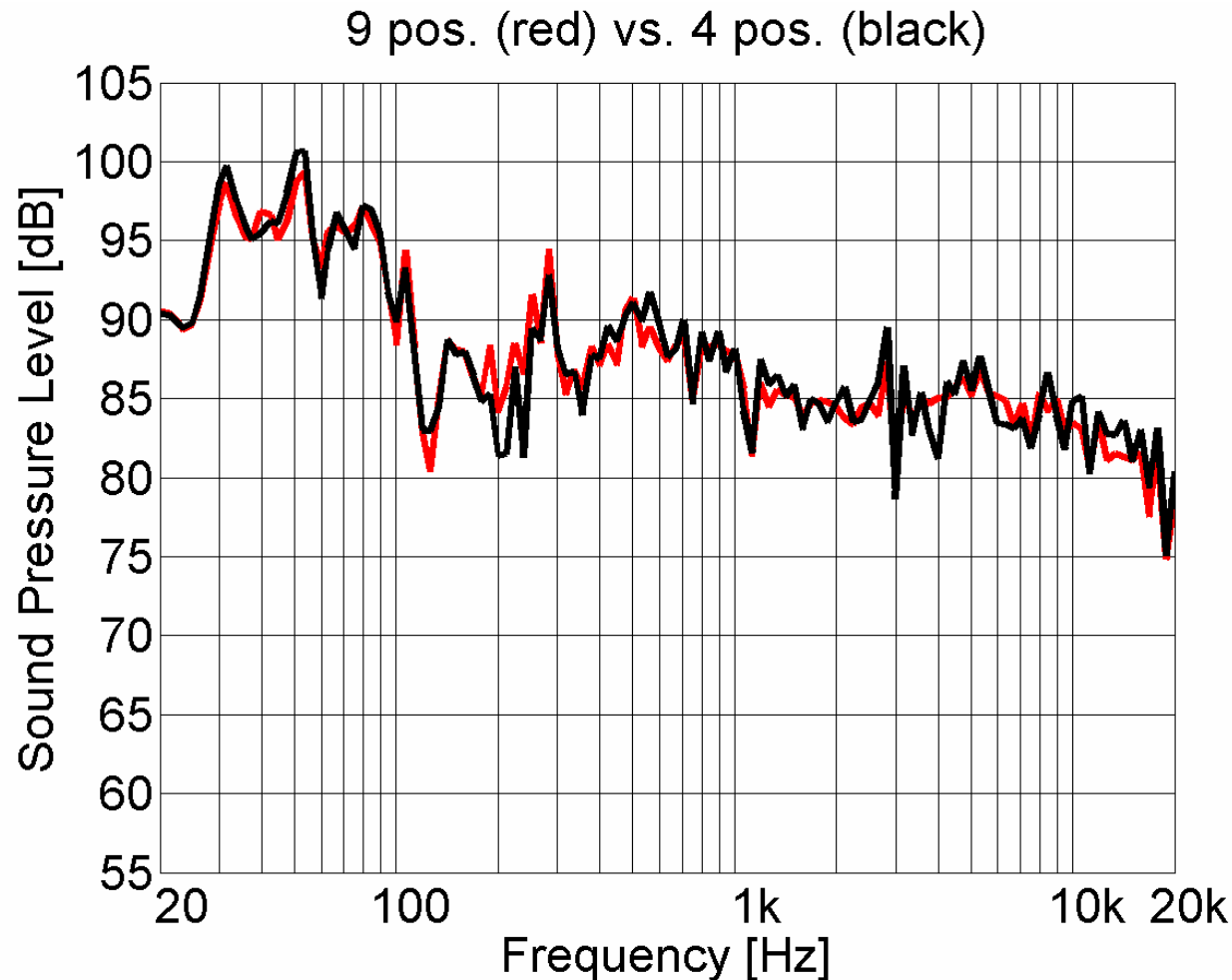
# Microphone Positions

## 3 Positions



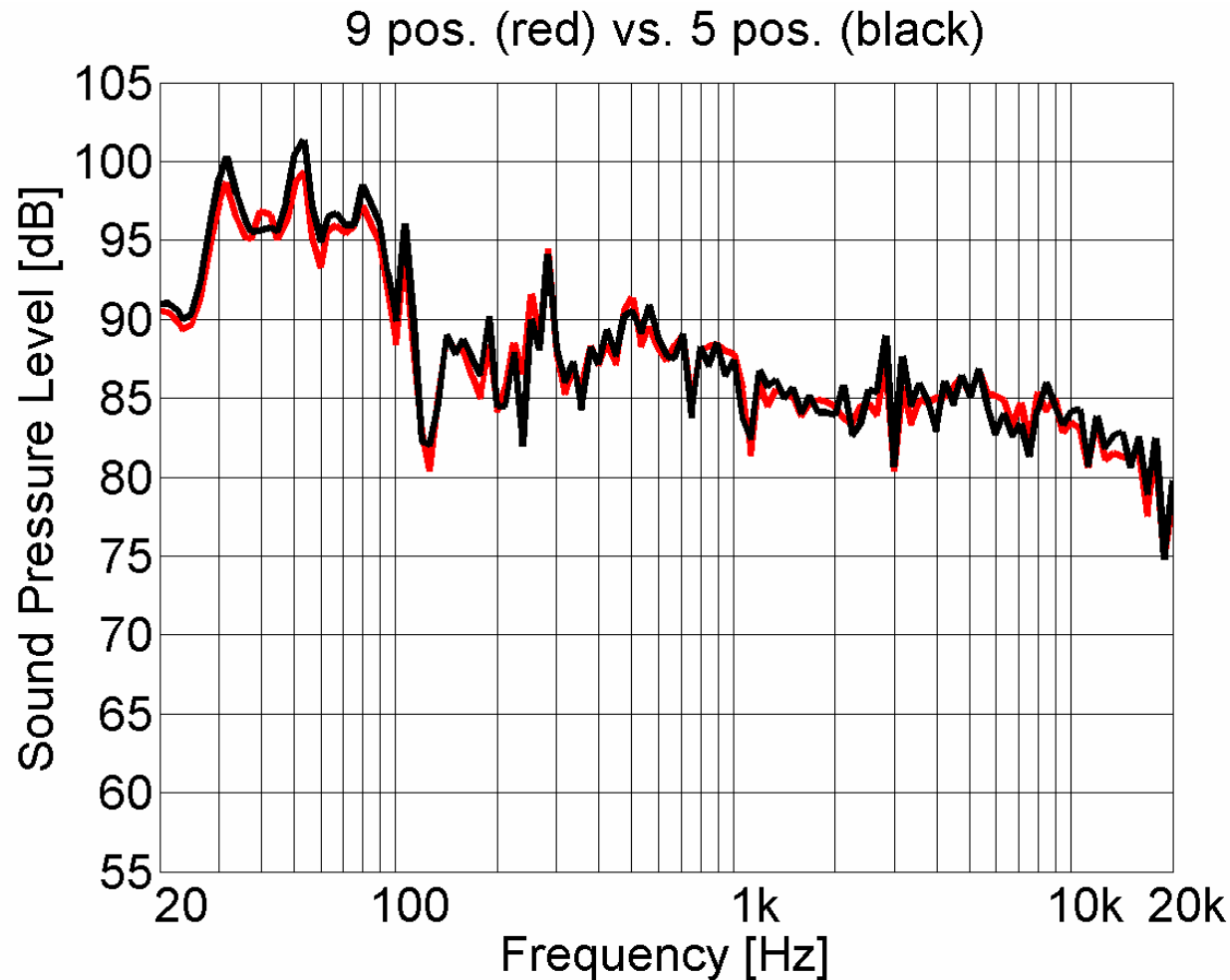
# Microphone Positions

## 4 Positions



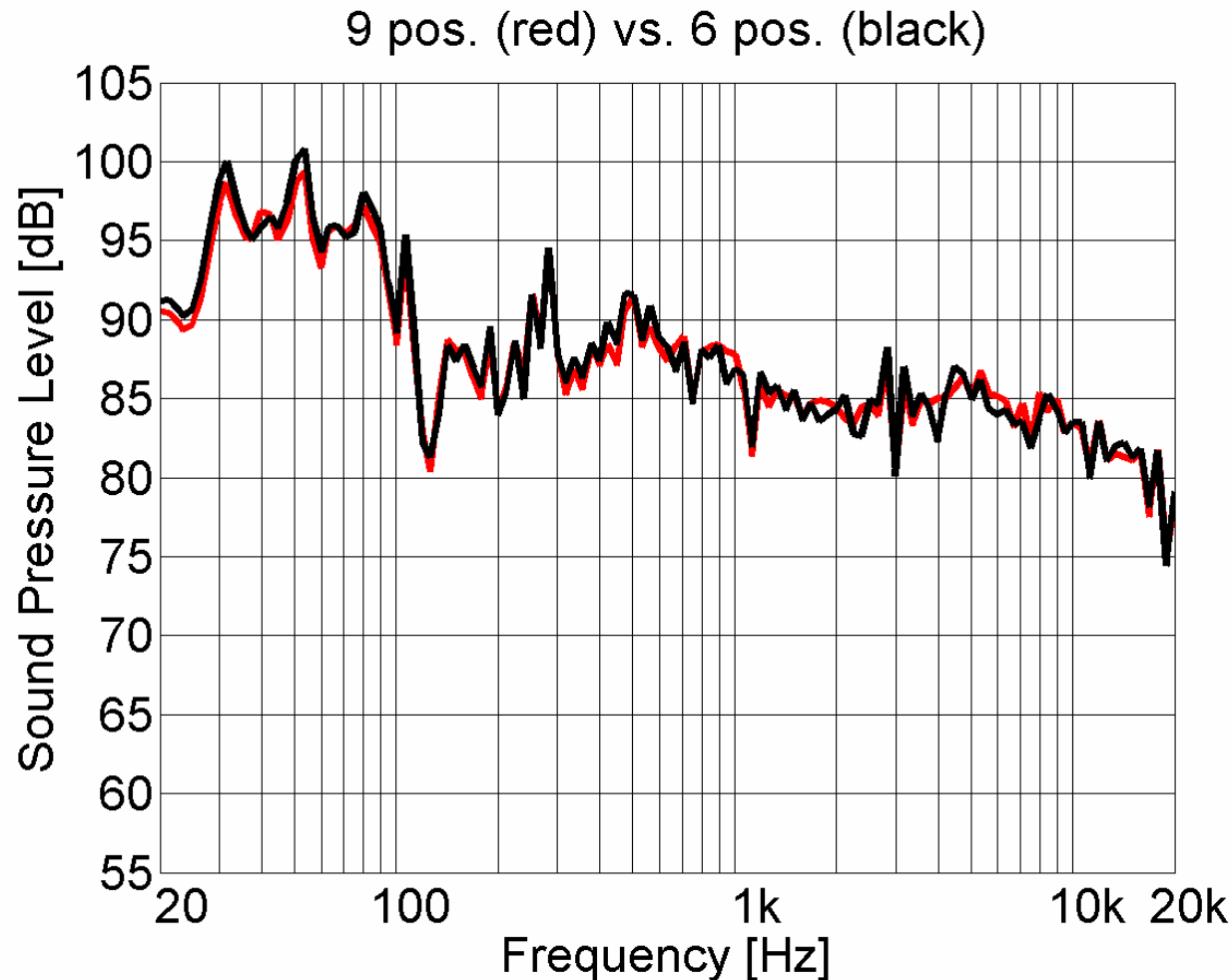
# Microphone Positions

## 5 Positions



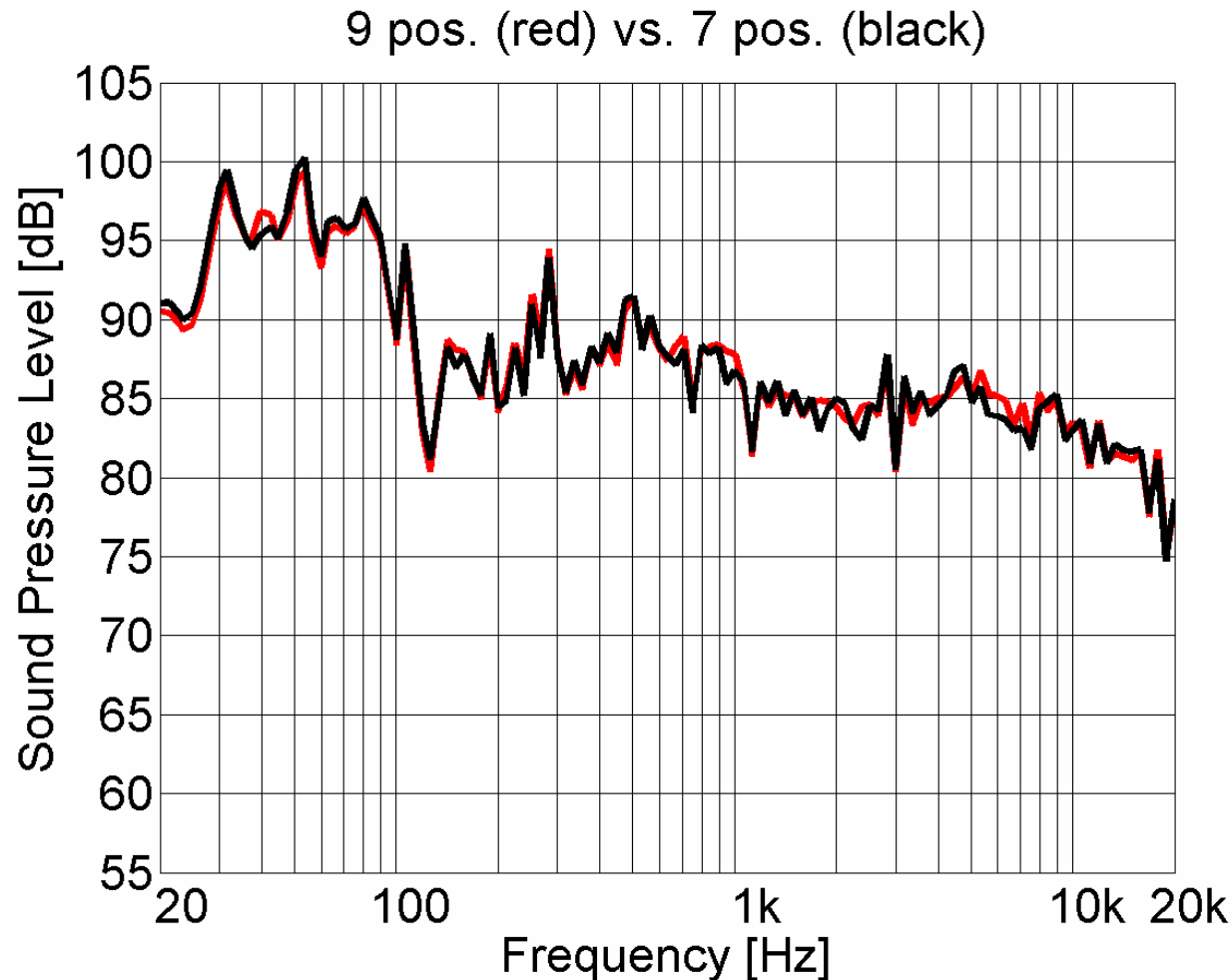
# Microphone Positions

## 6 Positions



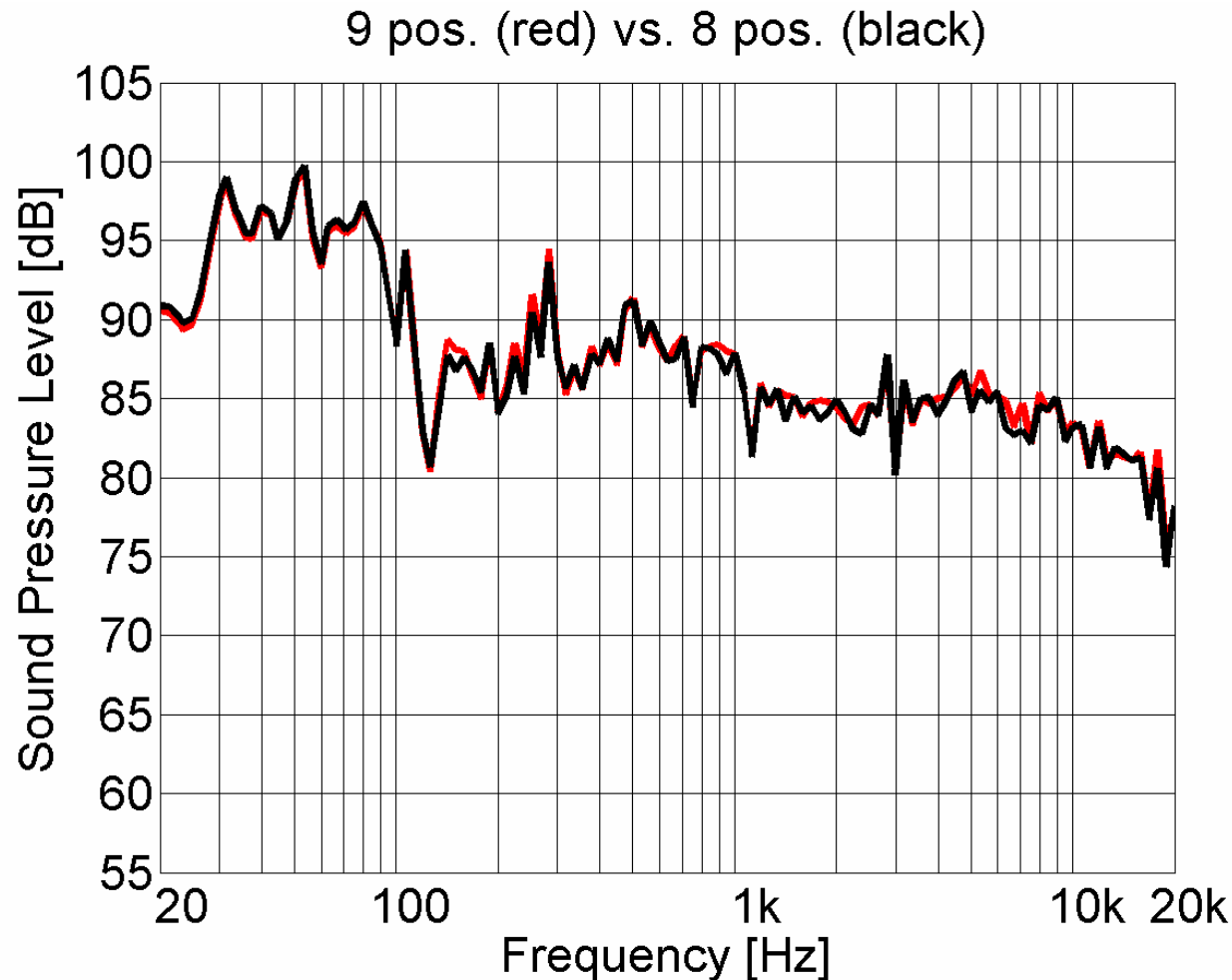
# Microphone Positions

## 7 Positions

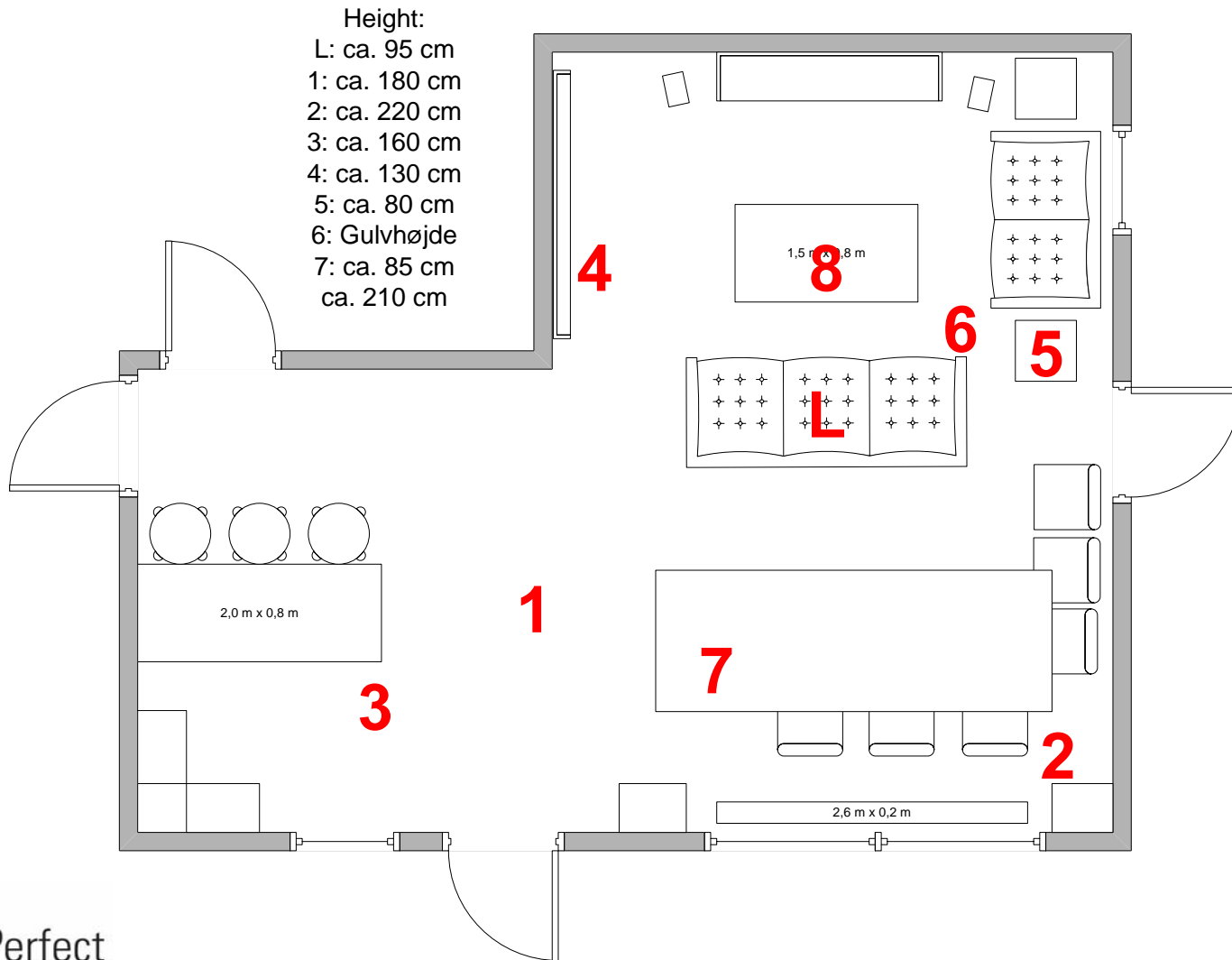


# Microphone Positions

## 8 Positions



# Measuring in a L shaped room

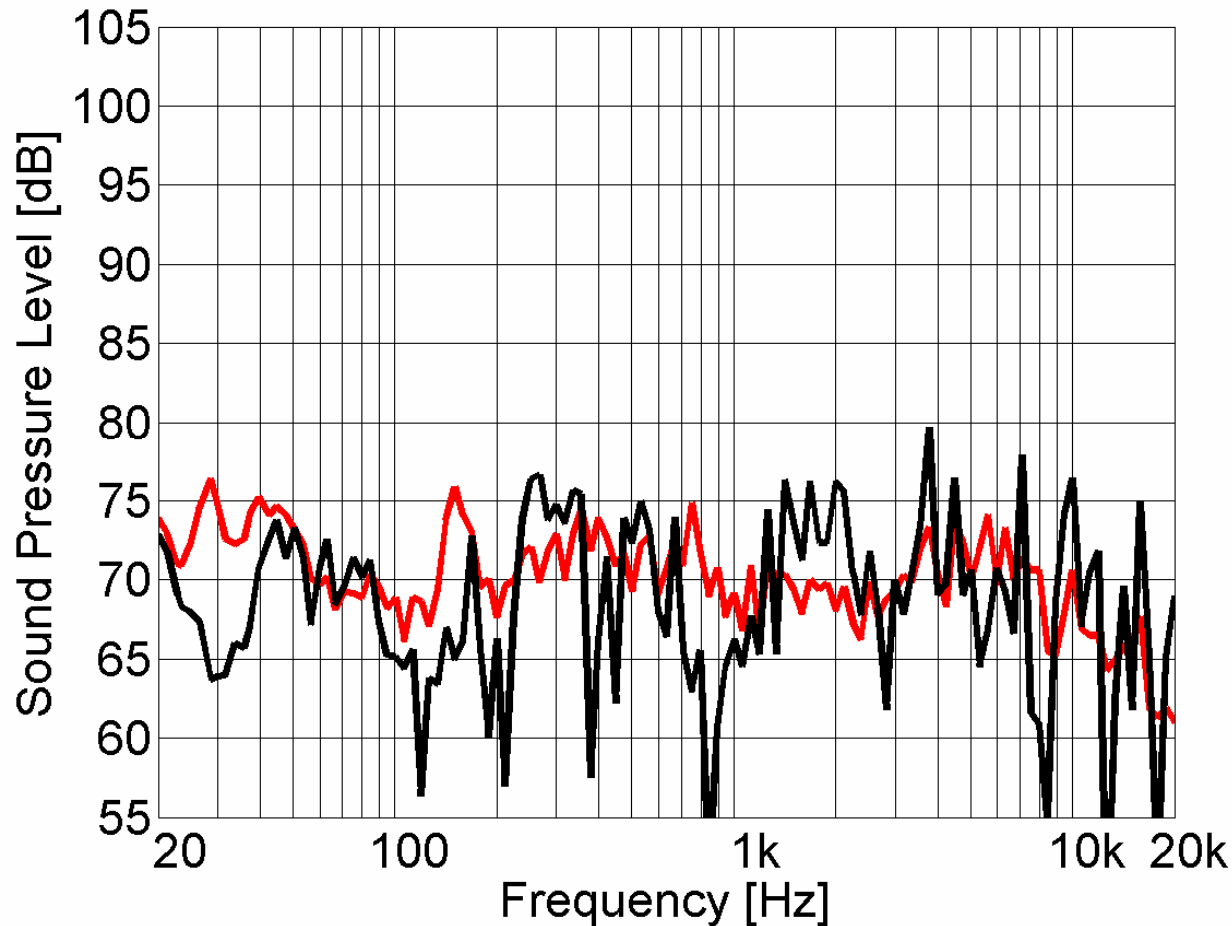




# Microphone Positions

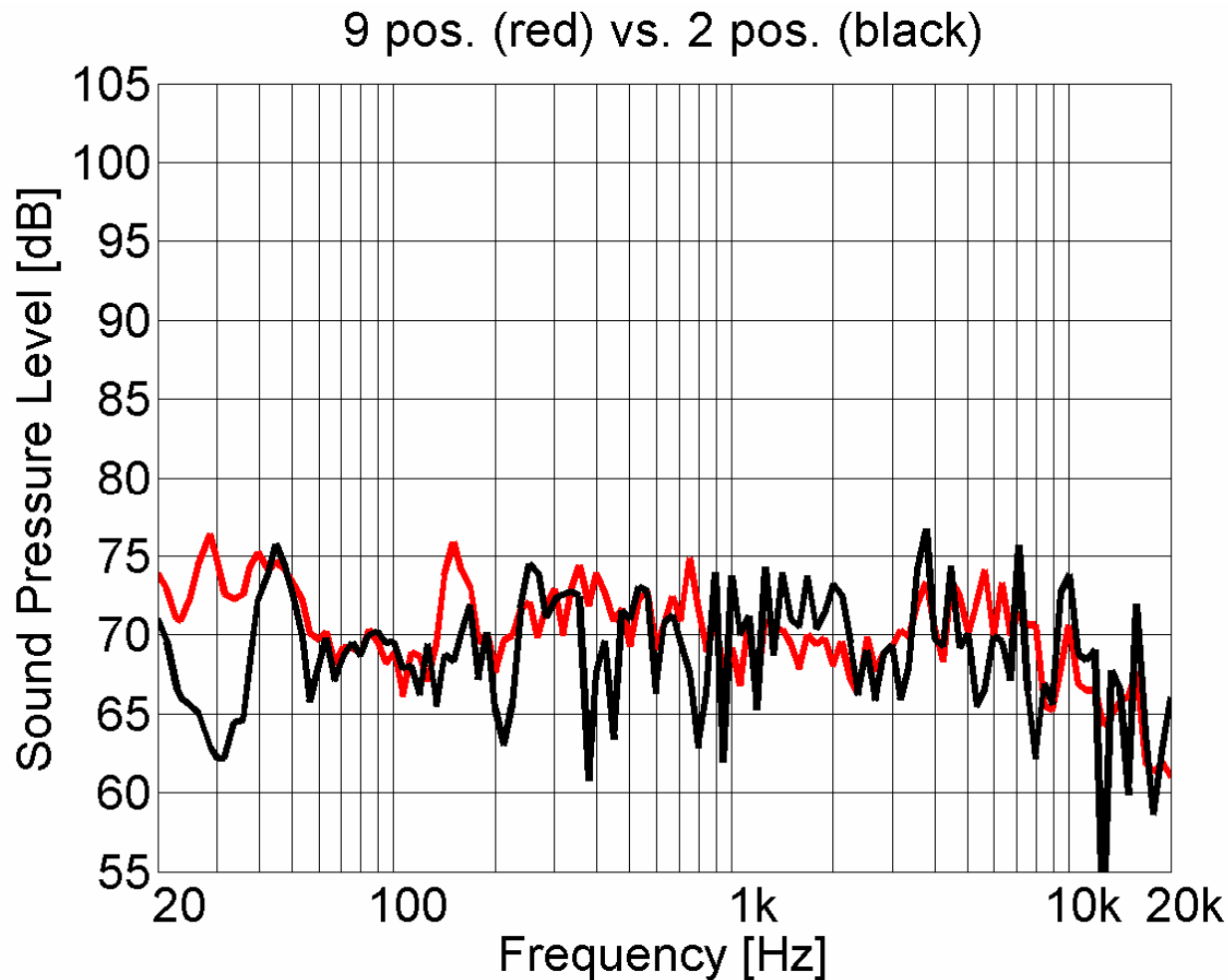
## 1 Position

9 pos. (red) vs. 1 pos. (black)



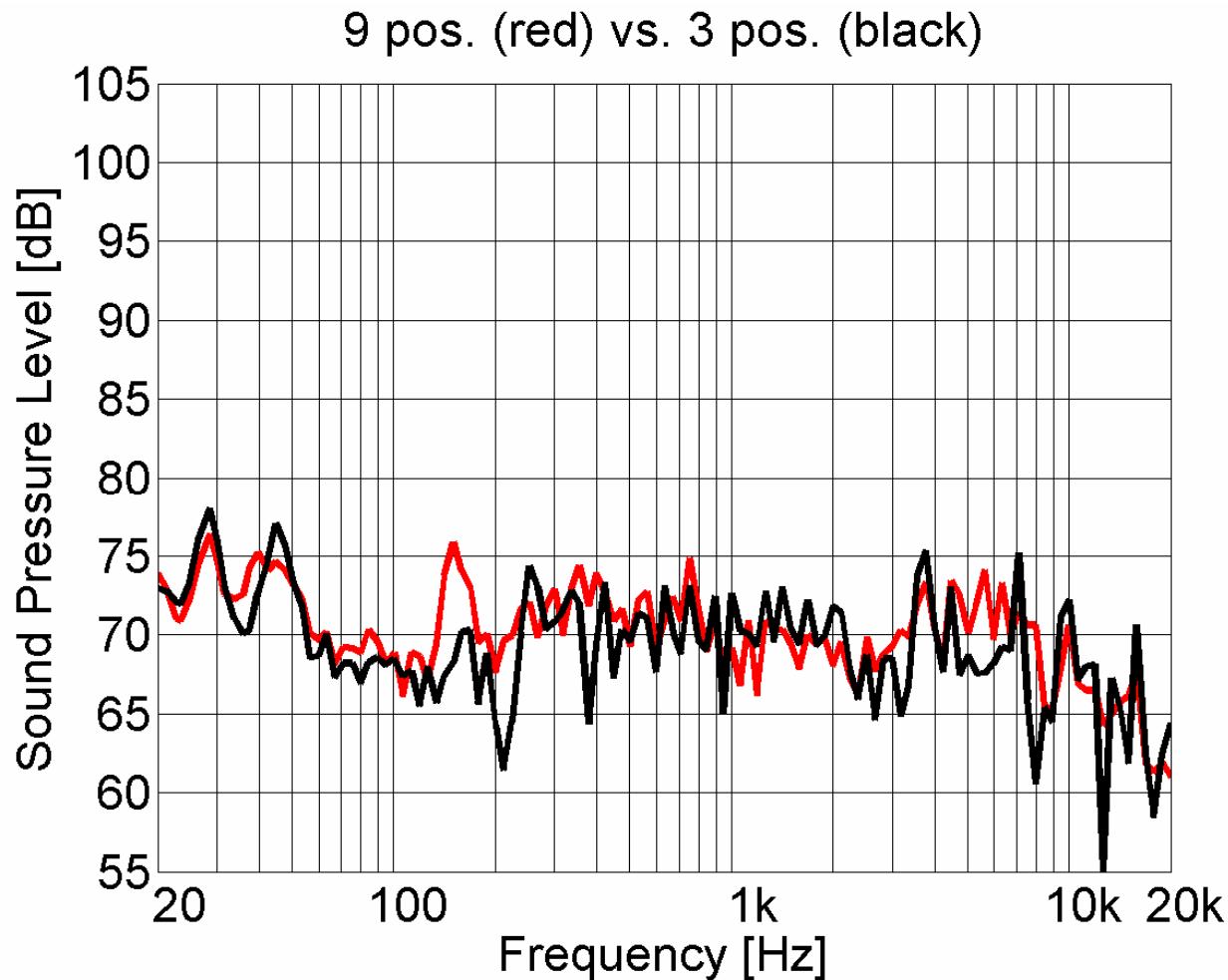
# Microphone Positions

## 2 Positions



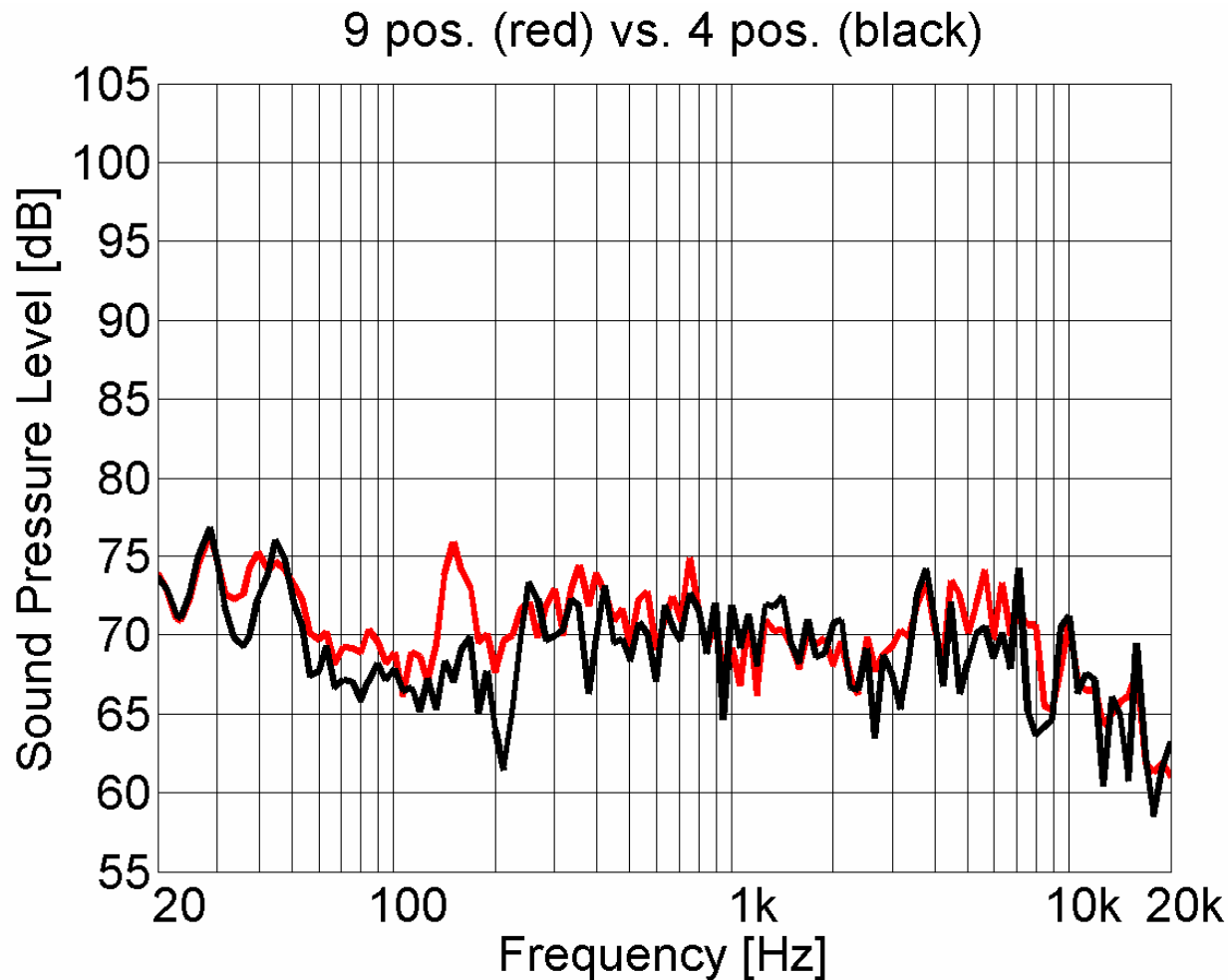
# Microphone Positions

## 3 Positions



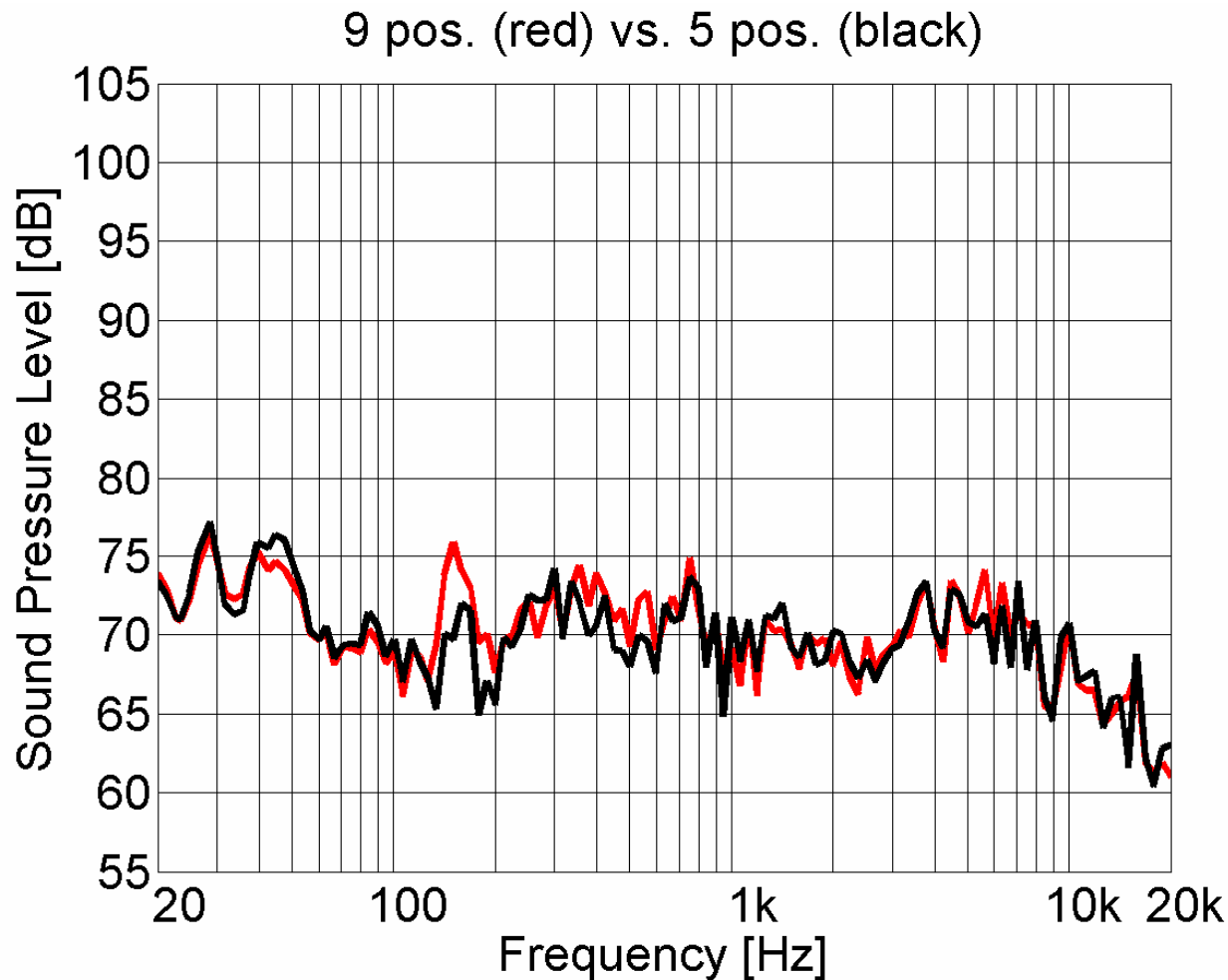
# Microphone Positions

## 4 Positions



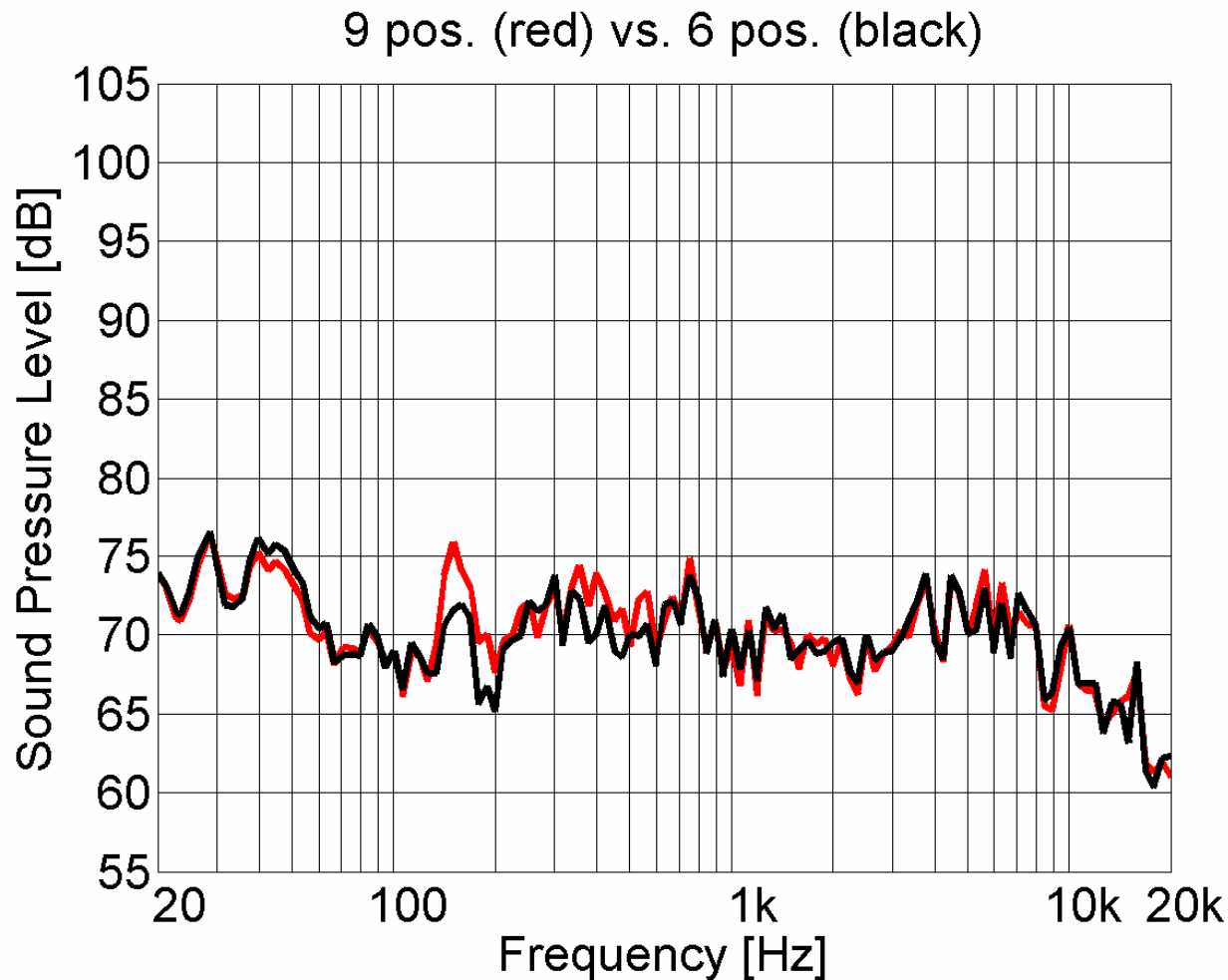
# Microphone Positions

## 5 Positions



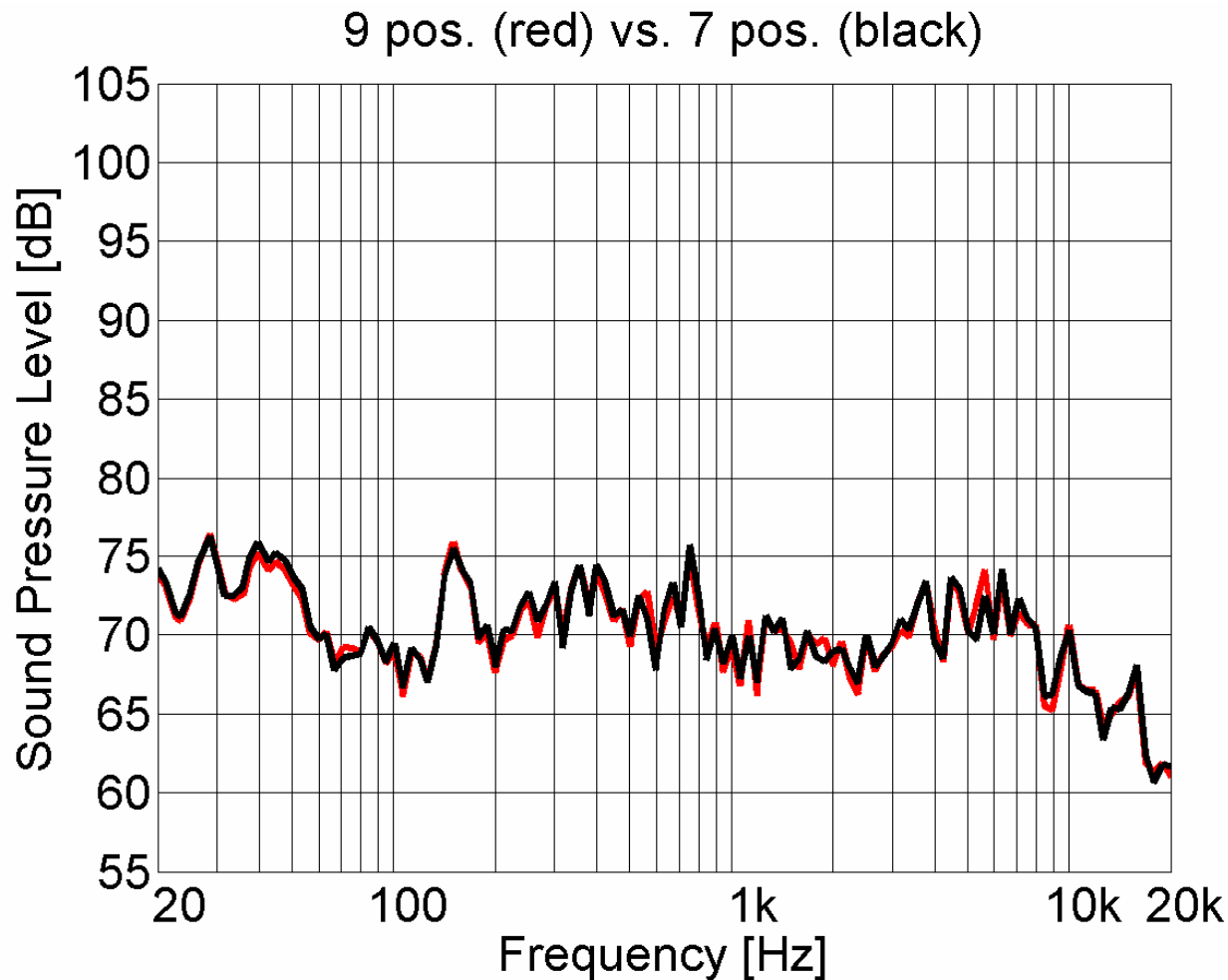
# Microphone Positions

## 6 Positions



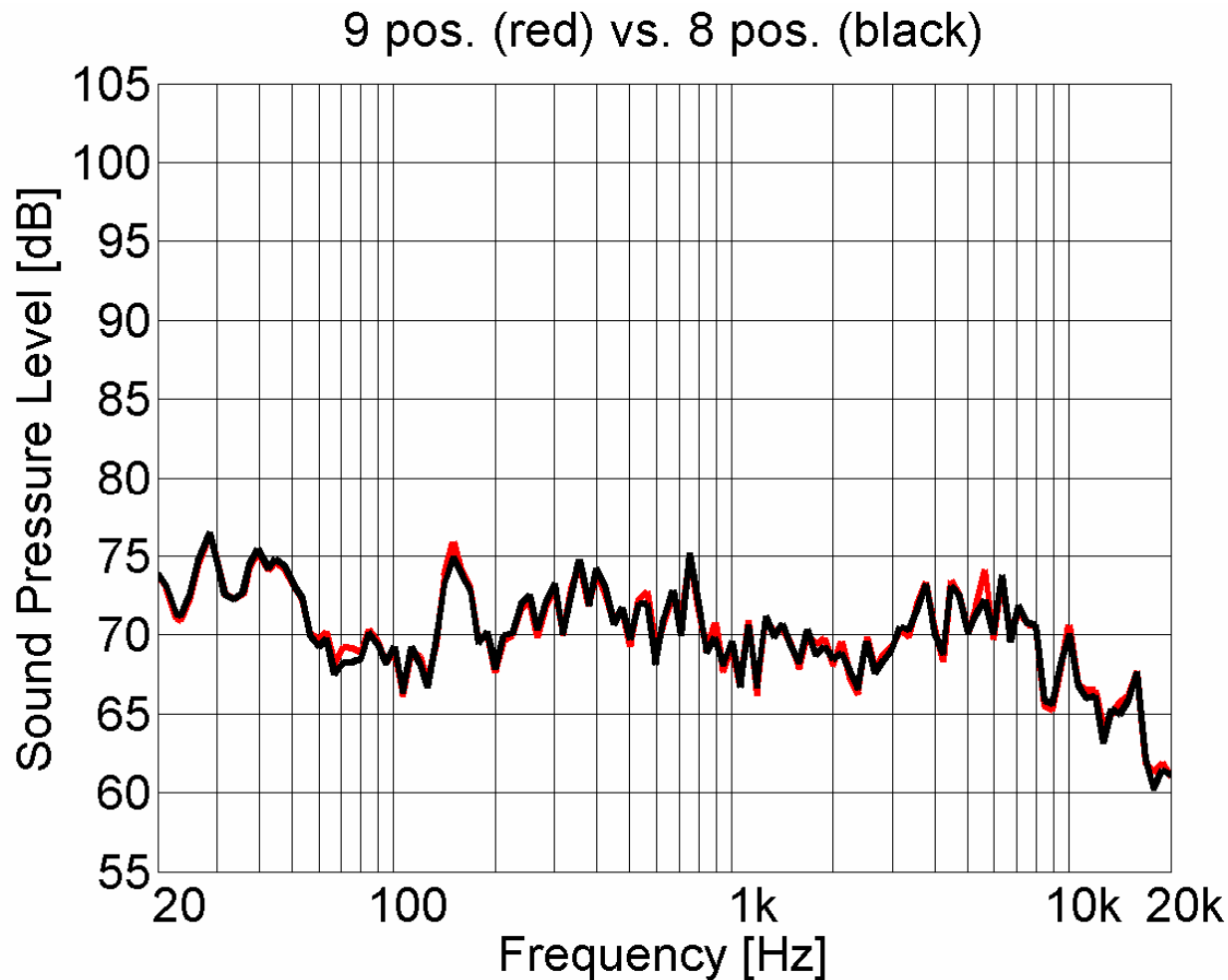
# Microphone Positions

## 7 Positions



# Microphone Positions

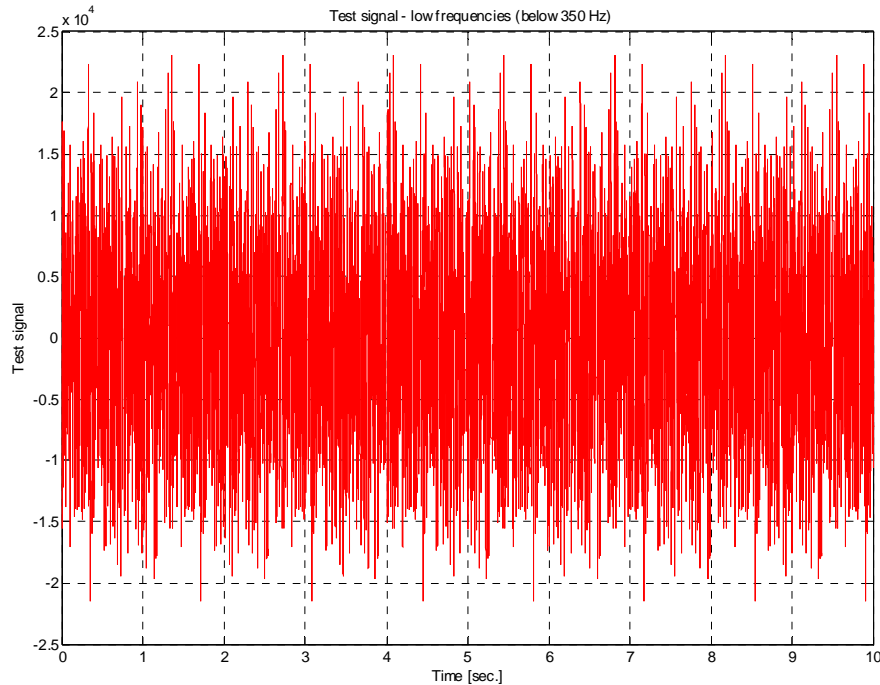
## 8 Positions



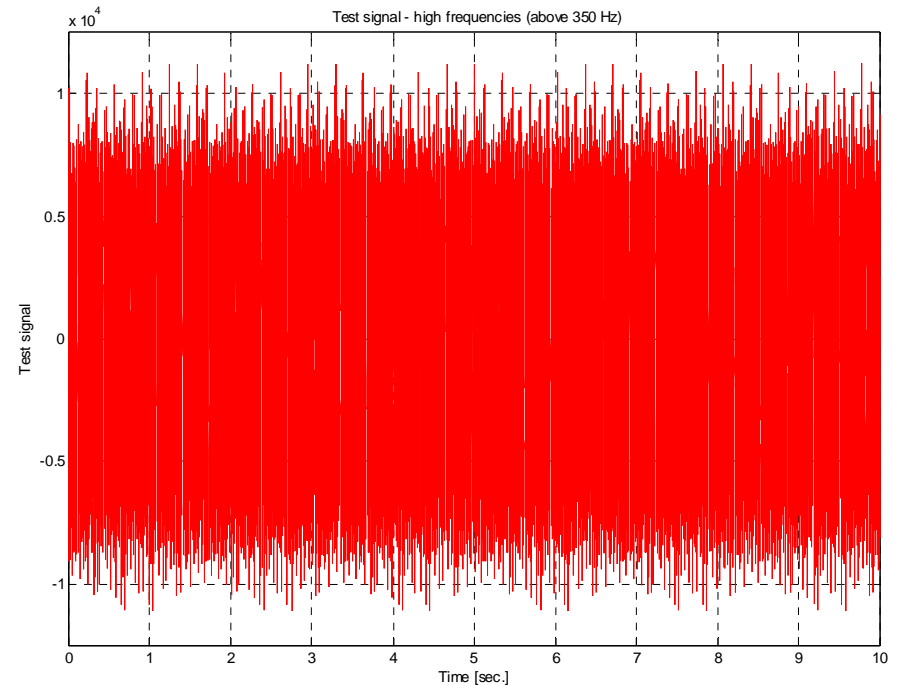


# Test Signal

## Multiple Pure Tones



Below 350 Hz (50 tones)



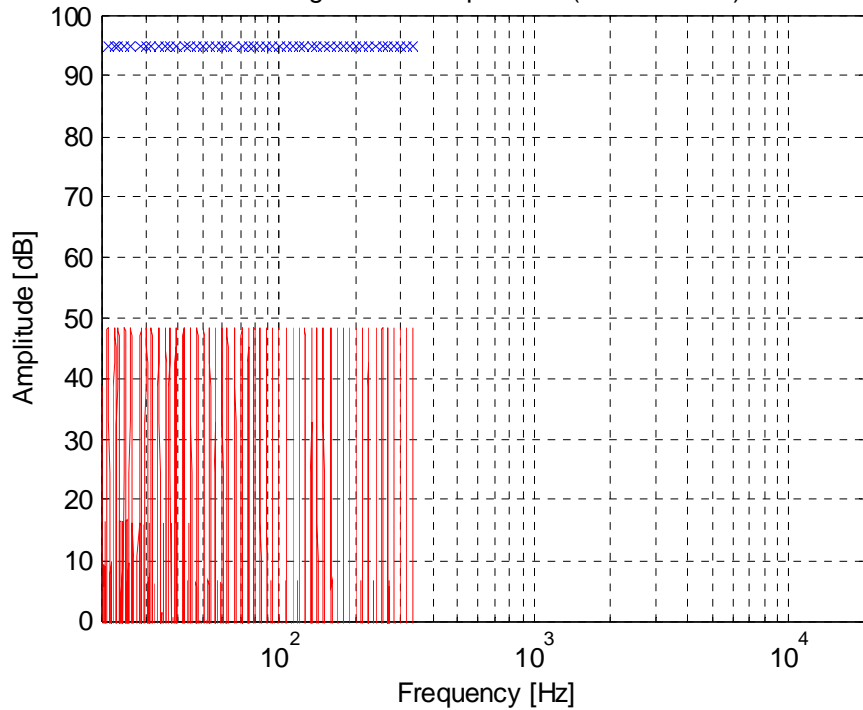
Above 350 Hz (71 tones)

# Test Signal

## Multiple Pure Tones

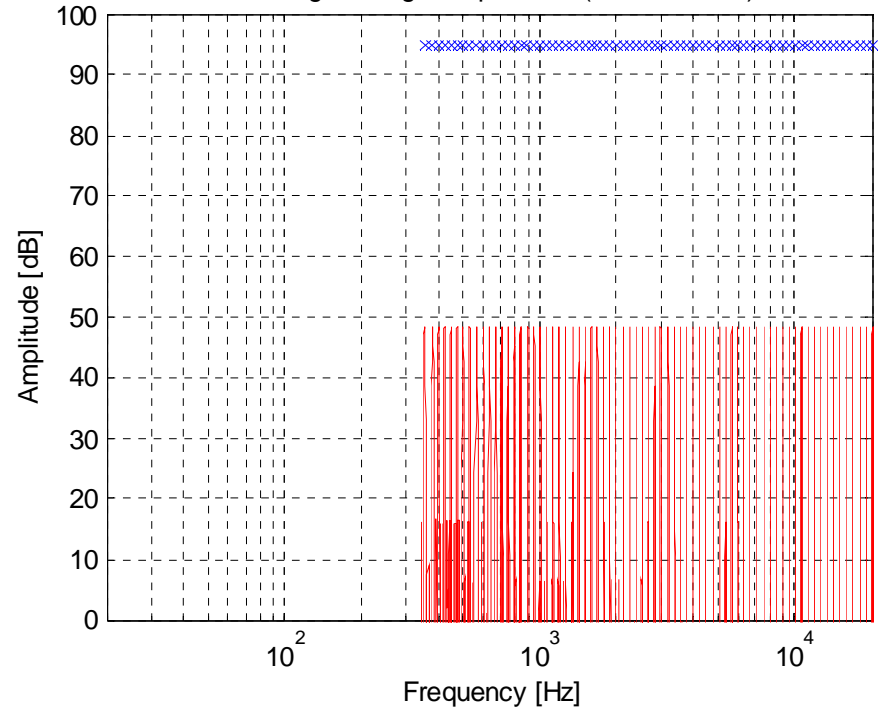
resolution depending on frequency – down to approx. 1 Hz @ 20 Hz

Test signal - low frequencies (below 350 Hz)



Below 350 Hz (50 tones)

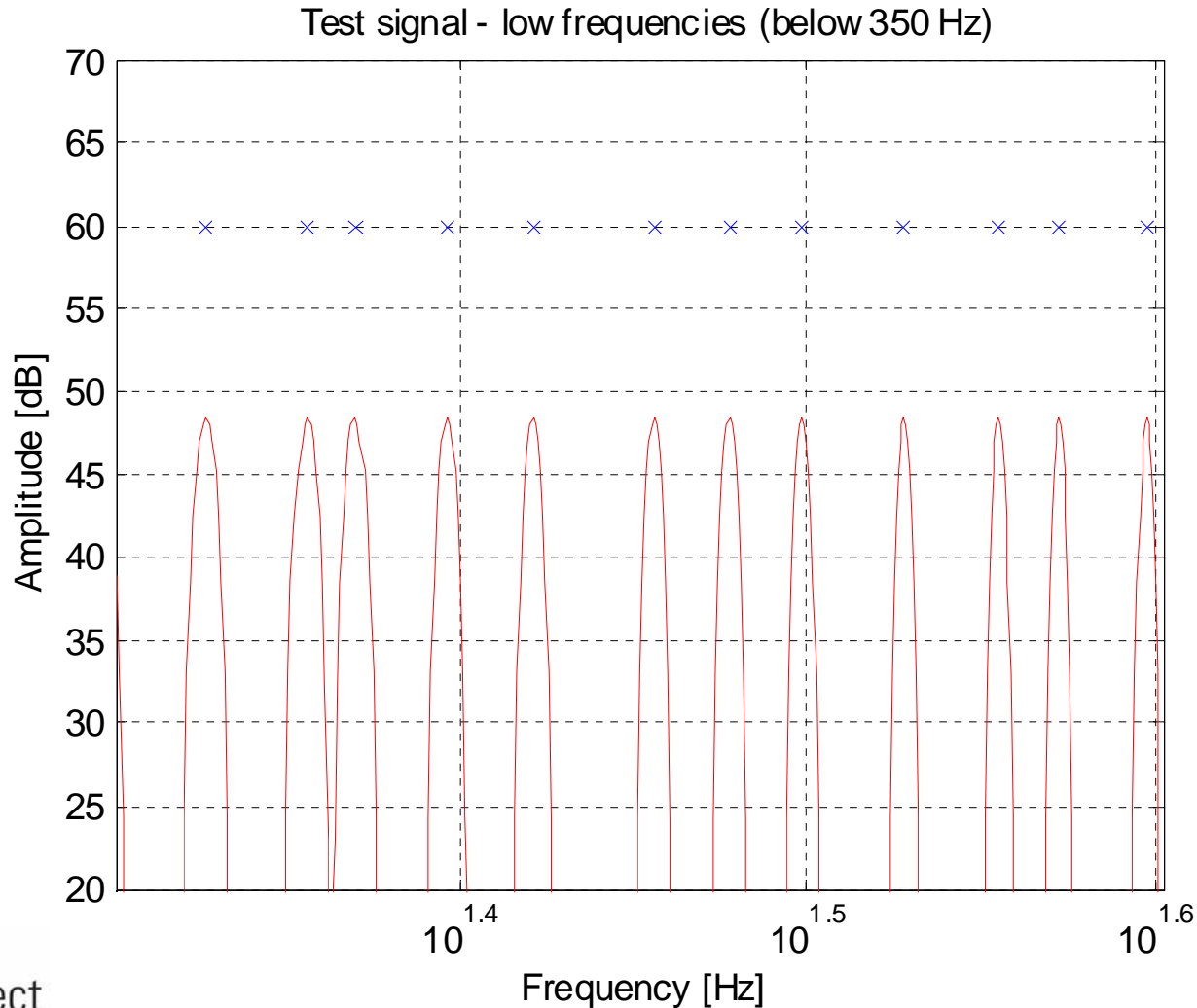
Test signal - high frequencies (above 350 Hz)



Above 350 Hz (71 tones)

# Test Signal

Pure Tones centered at the analysis frequencies filters out noise from the surroundings and through repeated calculations the measurement can be adjusted to any given precision



# Fully Automatic Target

New automated generation of target curve for the correction takes the characteristics of your loudspeaker into account:

Characteristics of lower cut off: limiting frequency and roll off order

-Sensitivity of the loudspeaker: the proper 0 dB level without room influence

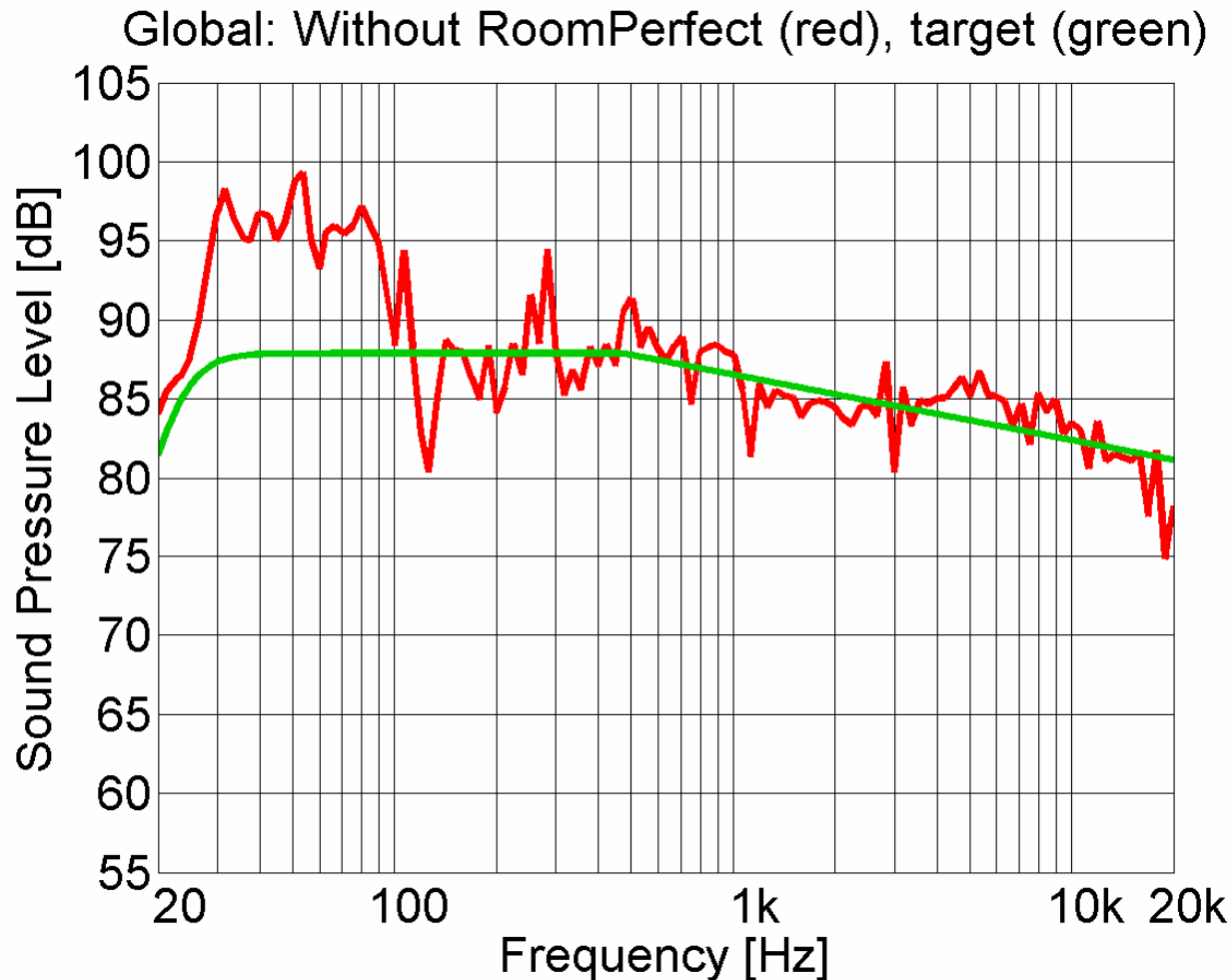
-Directivity characteristics: relationship between pressure response and power response

-A reasonable bass extension: a decreased lower cut off frequency as needed

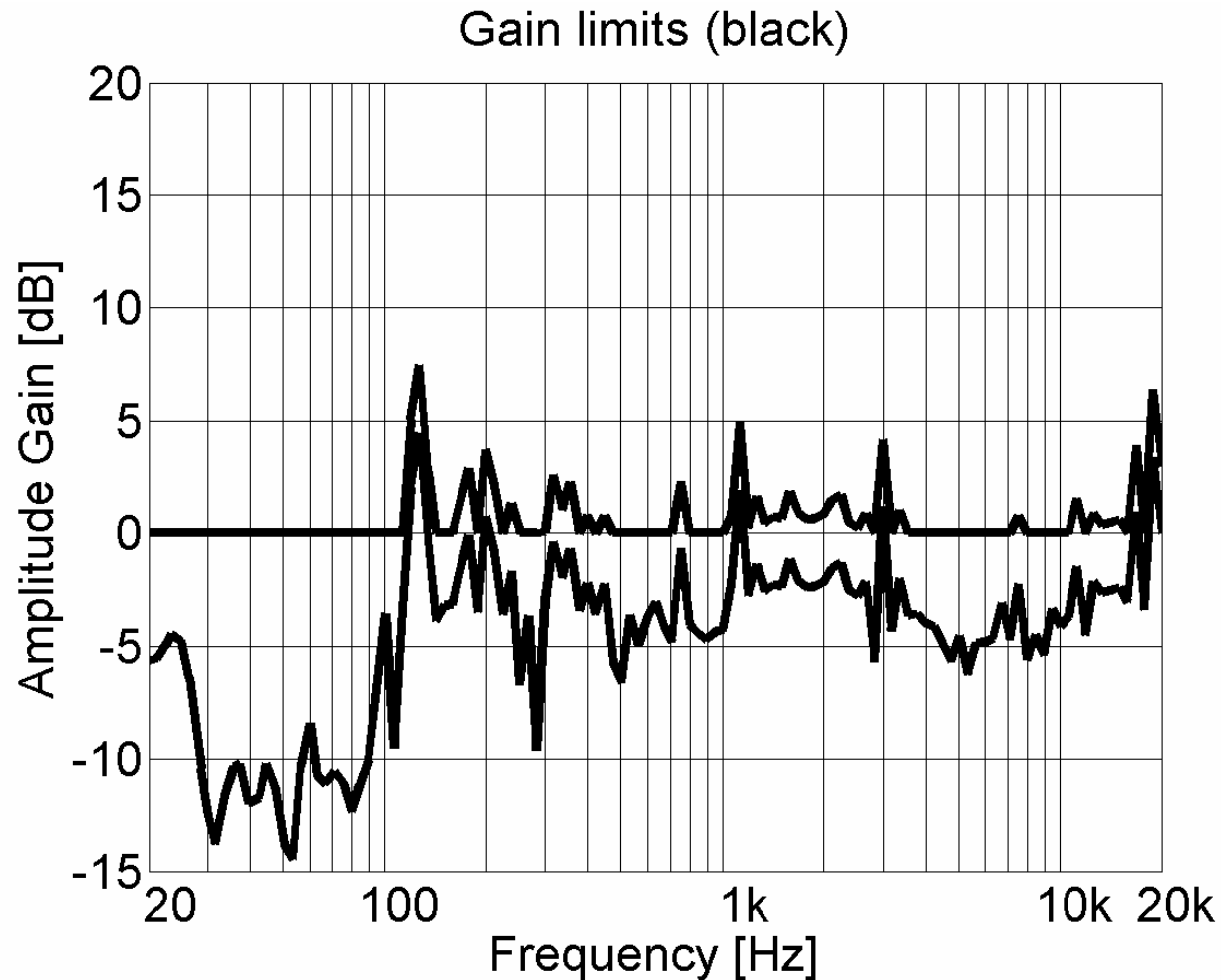
-No PC software needed – all functionality is embedded



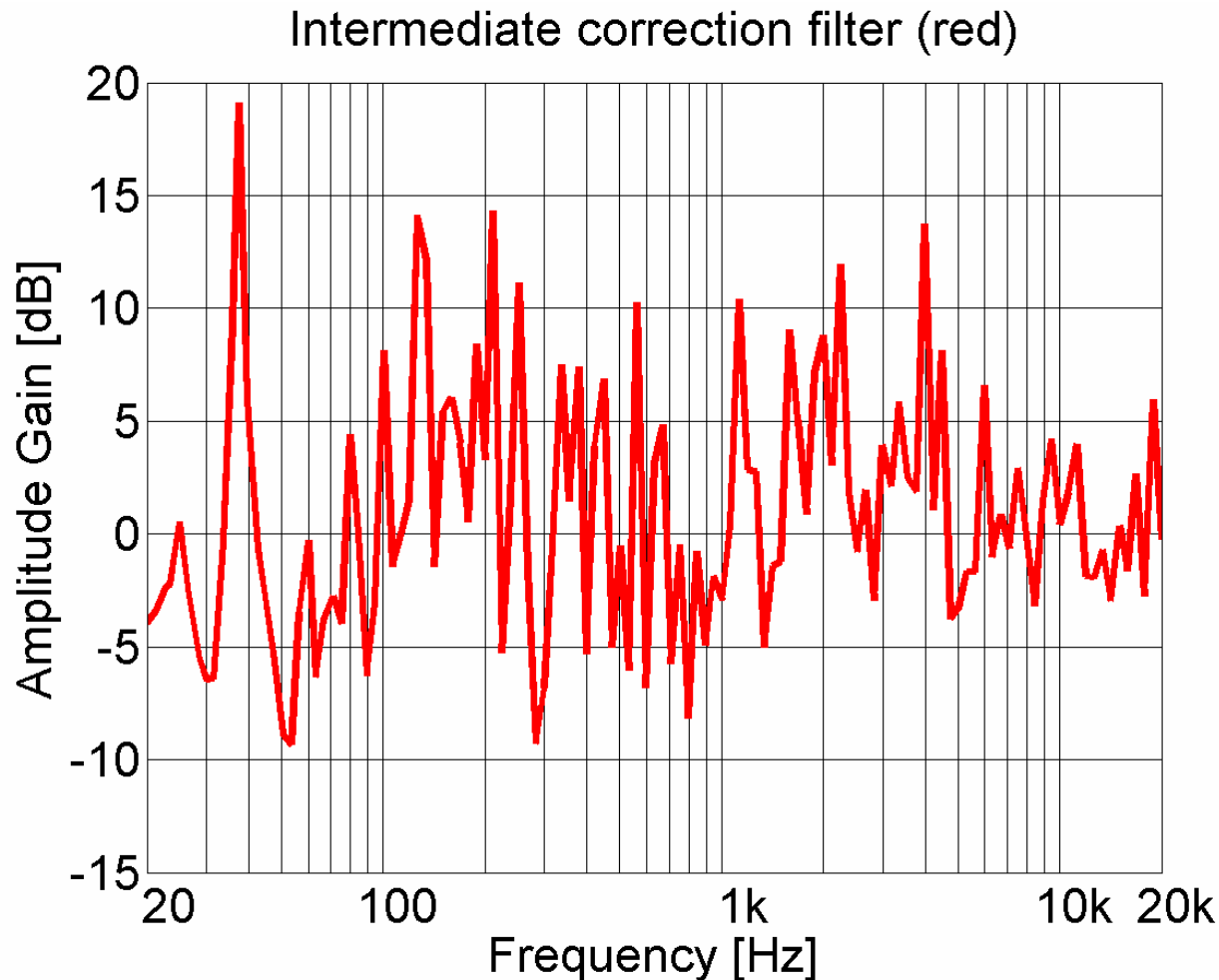
# Fully Automatic Target



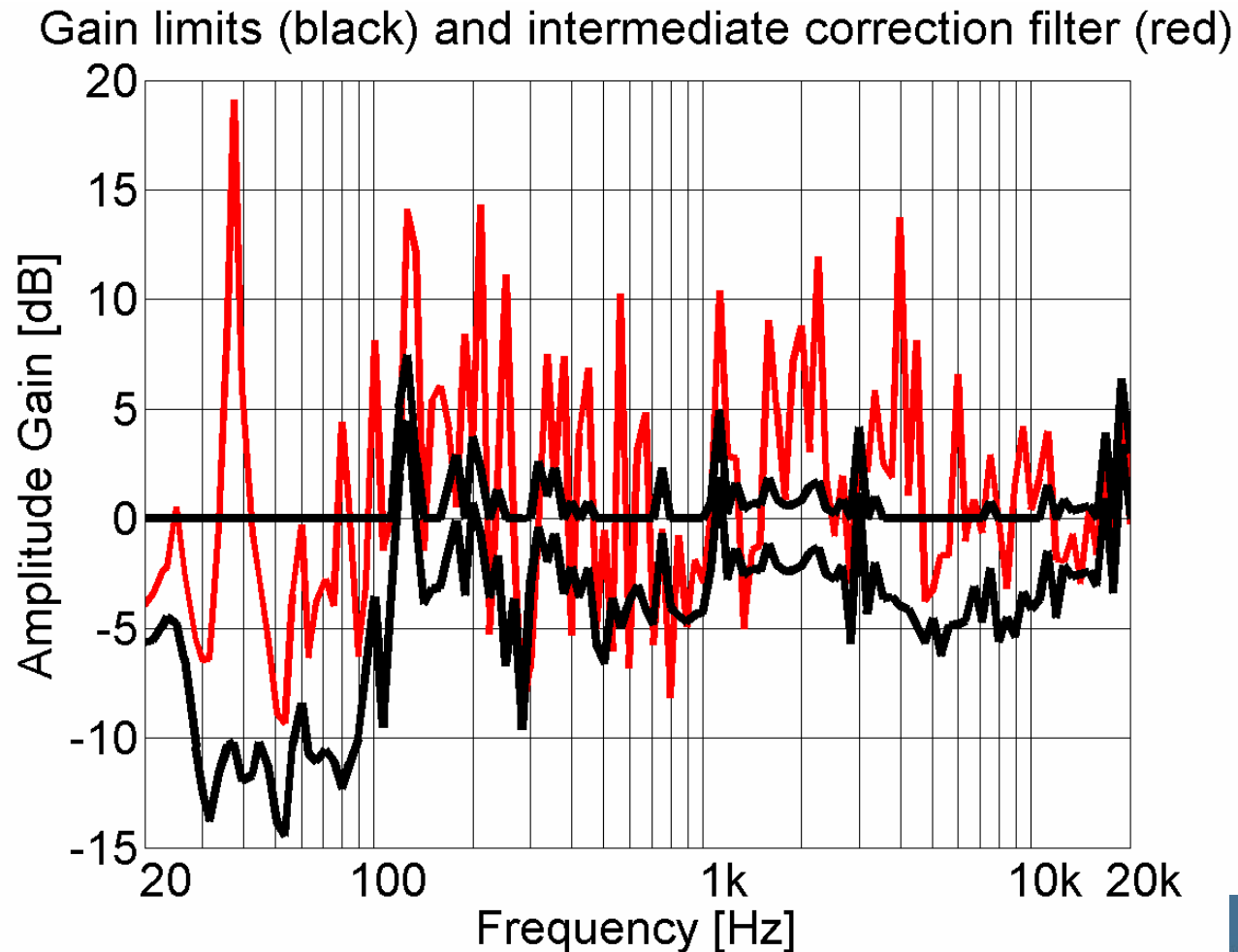
# Gain limits



# Unrestricted direct inverse

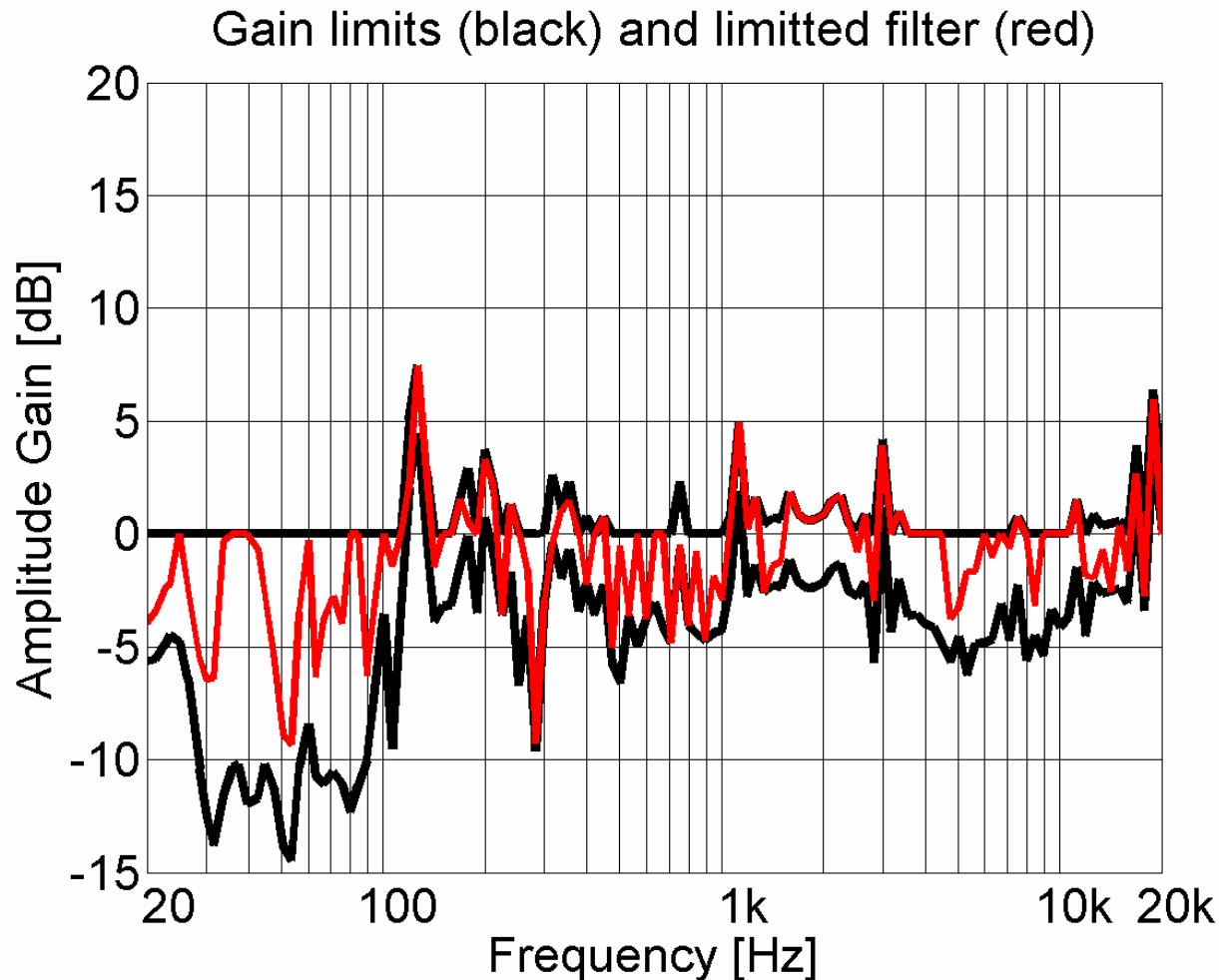


# Direct inverse and gain limits

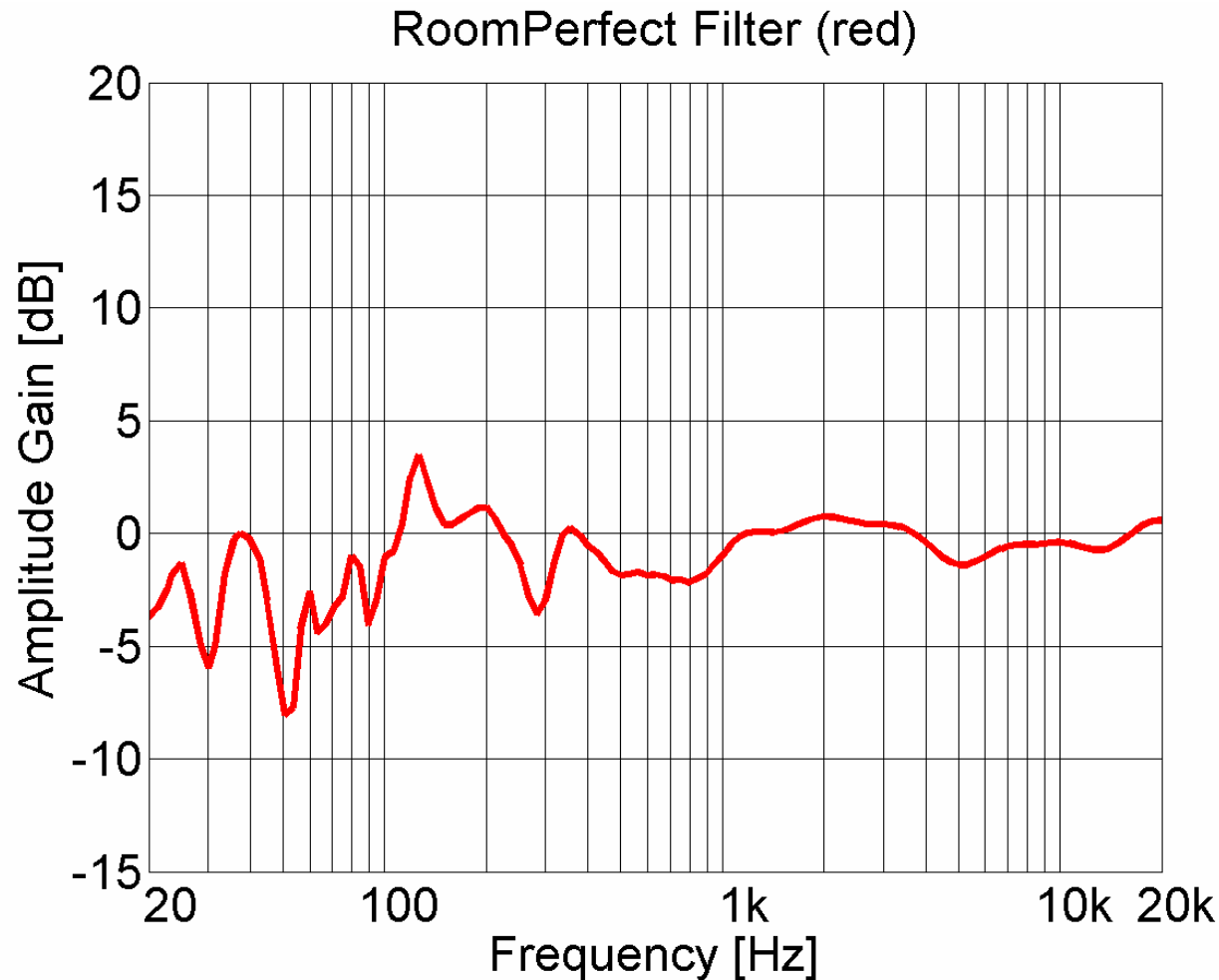




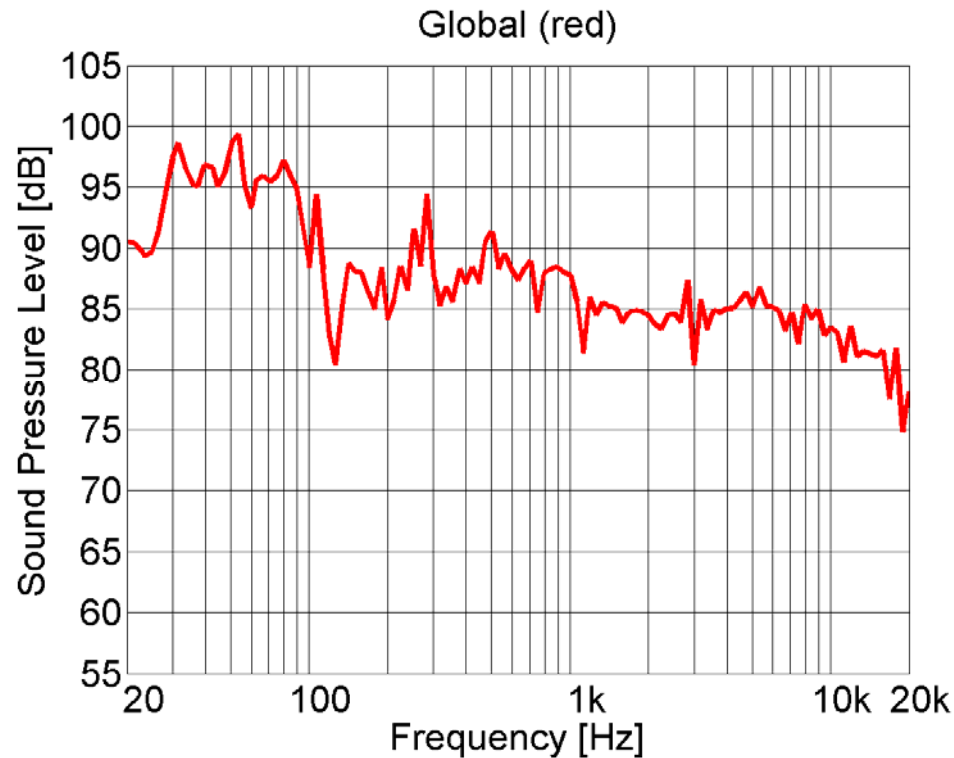
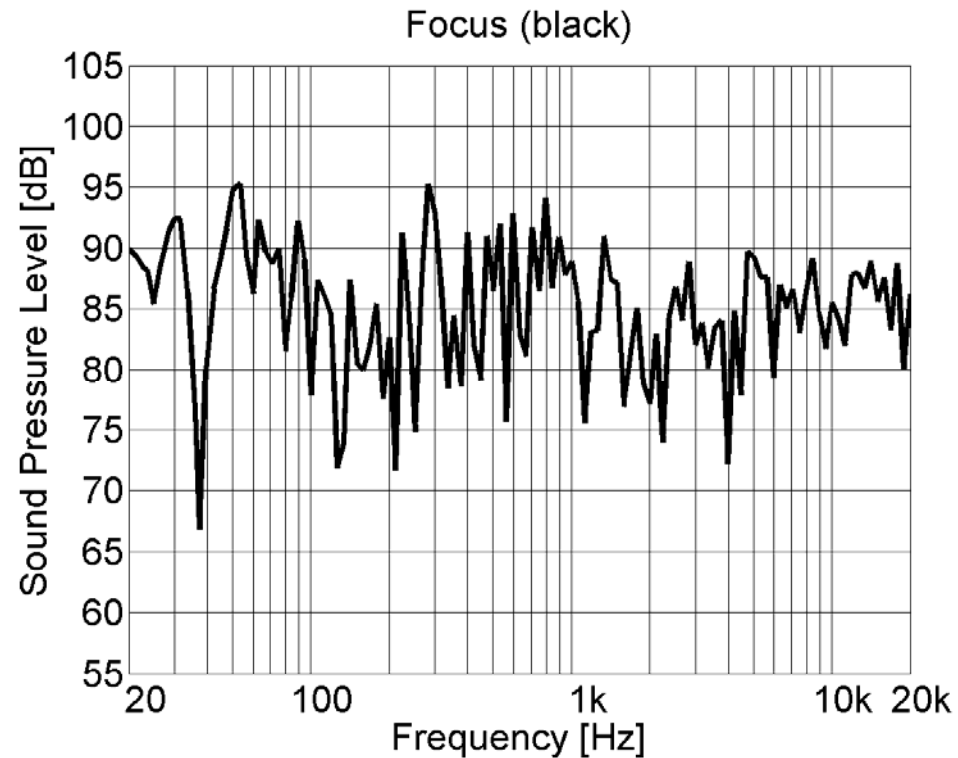
# Filter target after limitation



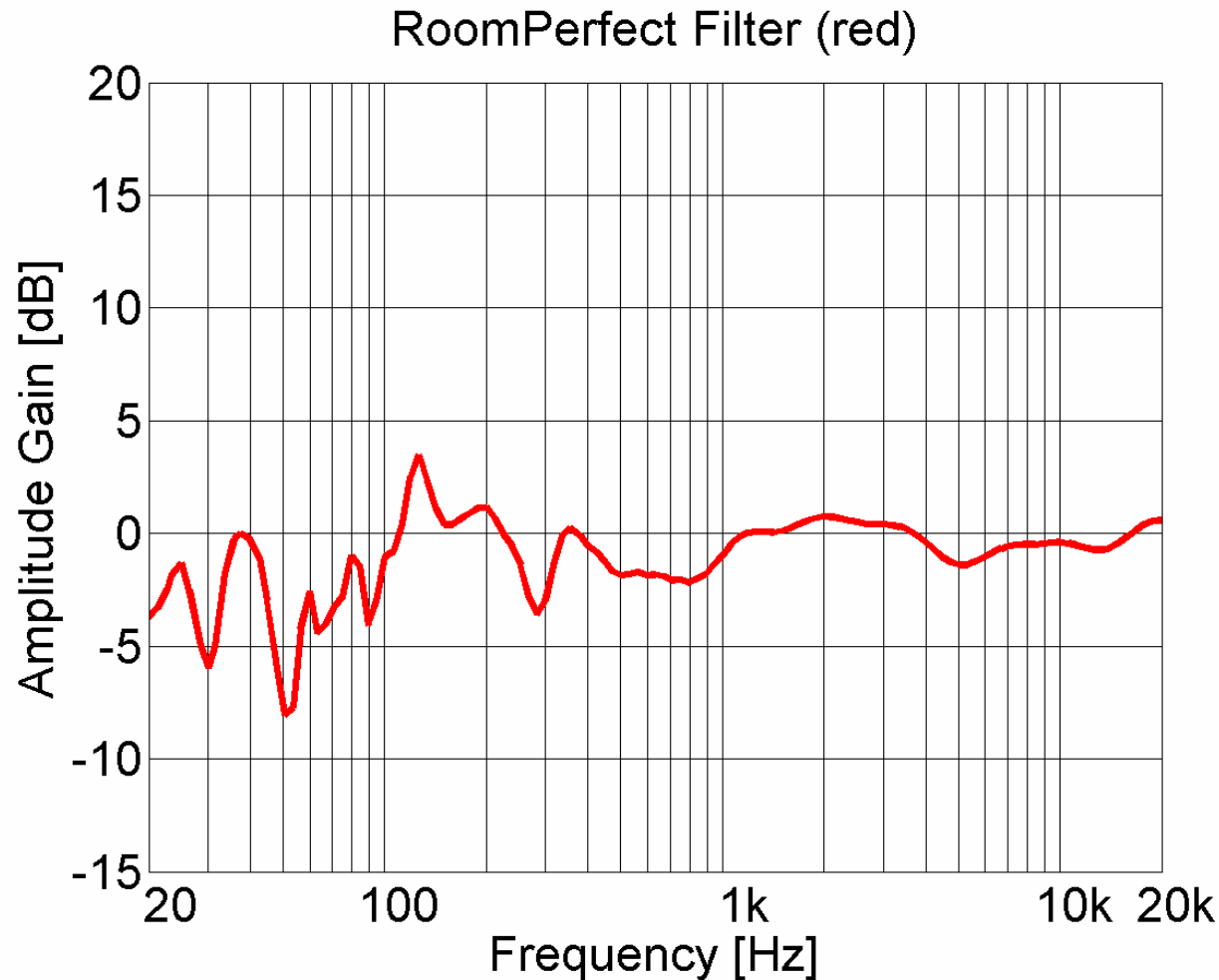
# Final correction filter



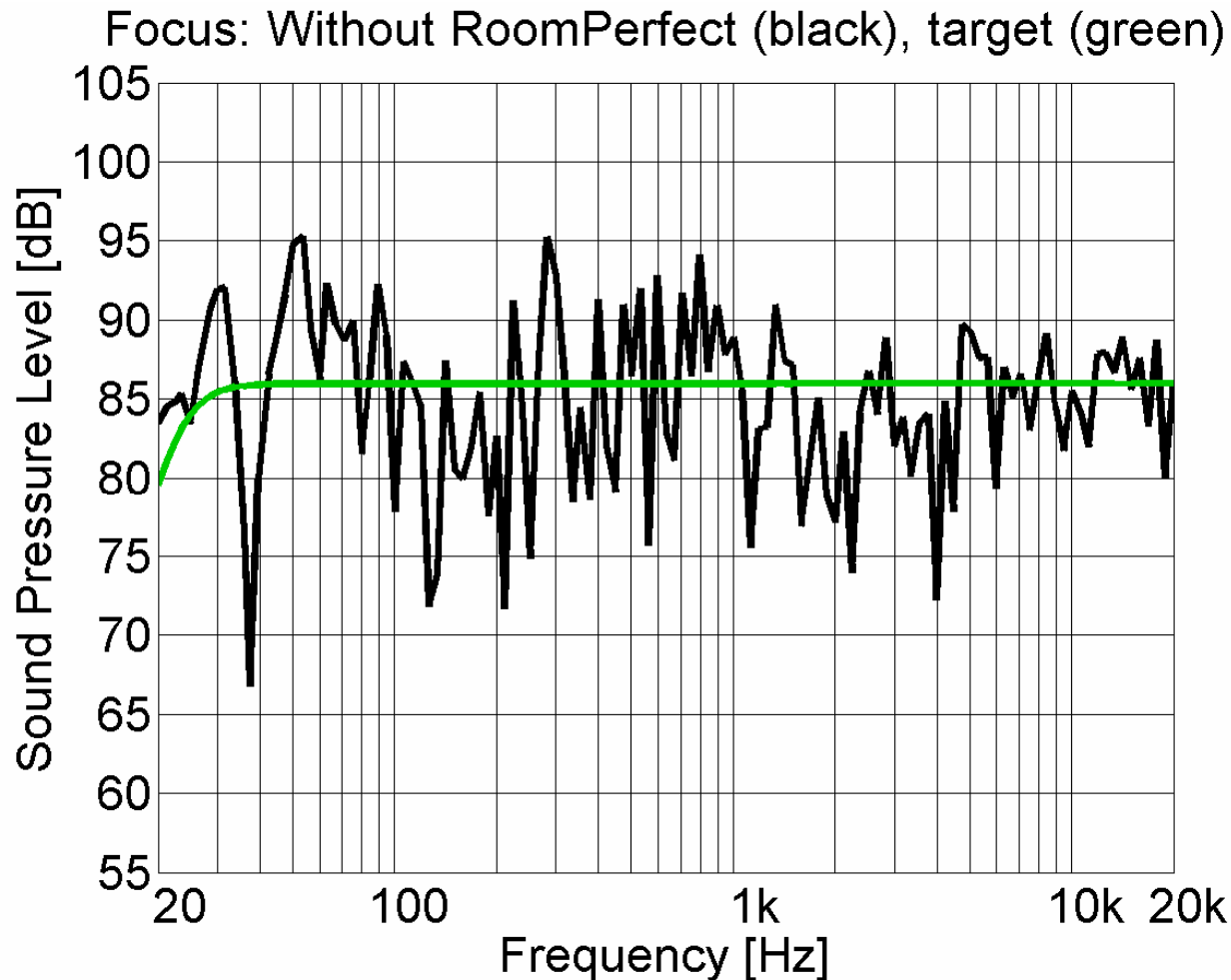
# Listening position vs. Global average



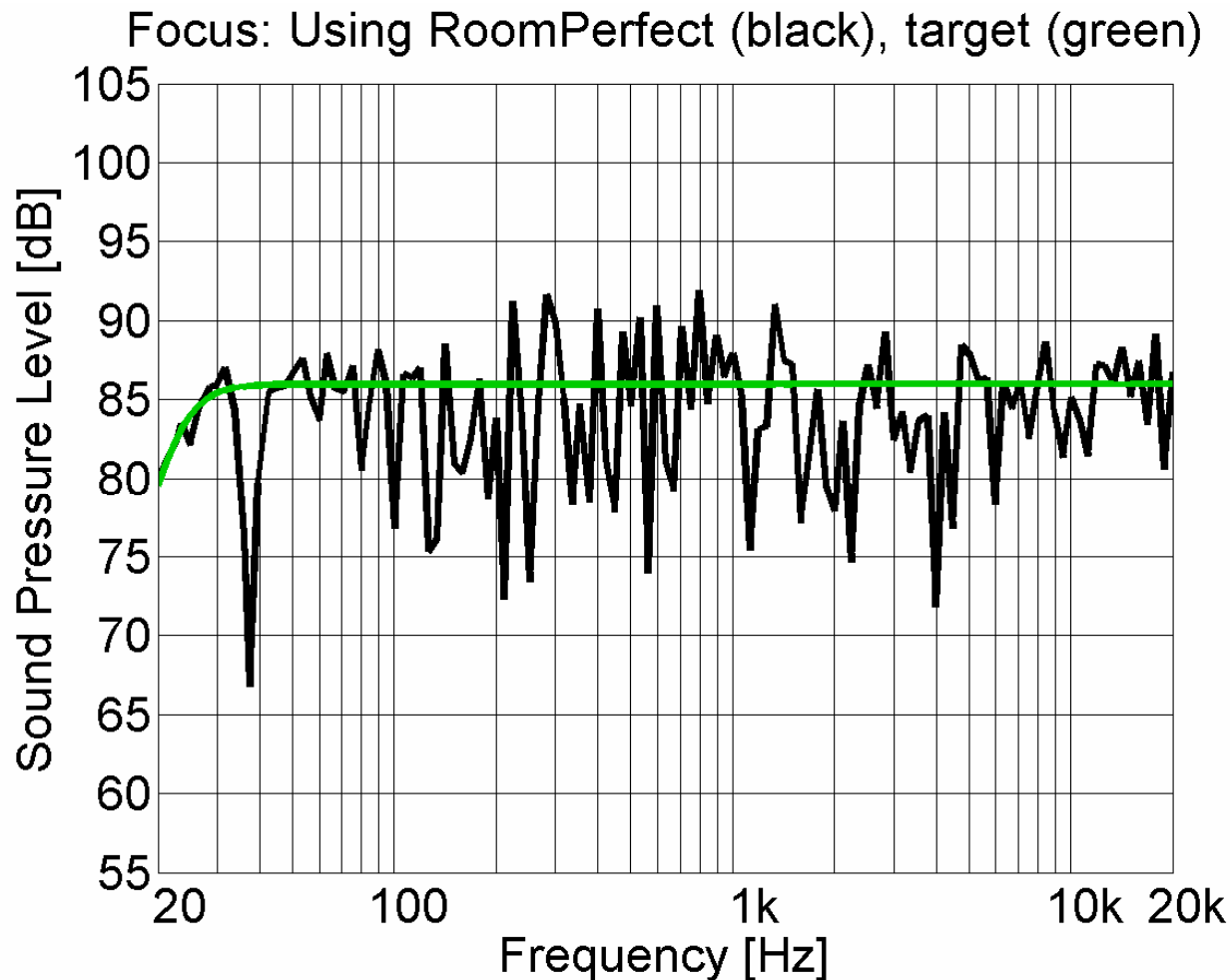
# Generated correction filter



# Listening position – before correction



# Listening position – after correction



# Conclusion

- Local optimization guided by the global characteristics of the listening room.
- Fully automatic target curve, which takes into account the properties of the actual used loudspeaker: lower cut-off frequency, directivity, sensitivity and treble cut-off frequency.
- Sampling of the energy in the 3D sound field using random microphone positions.
- Overcomming the problems of direct inversion – local uncontrolled high amplitude gain.
- Method does not rely on frequency smoothing.
- Acoustics and DSP based method for room adaptation.

Audio Engineering Society

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## Loudspeaker-Room Adaptation for a specific Listening Position using Information about the Complete Sound Field

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### ABSTRACT

A novel method is presented for equalizing a loudspeaker, for a specific listening position in order to compensate for an influence of the room in which it is positioned. The method is based on measuring the sound pressure in the listening position (focus position) and in at least 3 randomly selected positions scattered across the entire listening room (room positions). The measurement in the listening position holds information about the listener's access to the sound field while the room positions hold information about the energy in the 3D sound field. The correction for the listening position is then bound by upper and lower gain limits calculated as a function of frequency from the information about the 3D sound field.



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