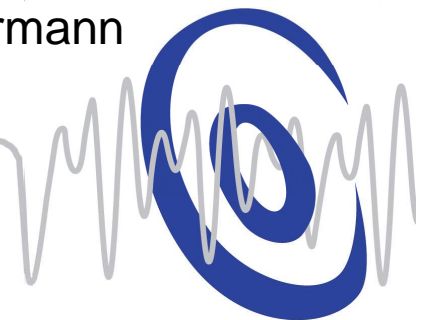


Perceptual Evaluation of Signal Enhancement Strategies

Koen Eneman, Heleen Luts, Jan Wouters, Michael Büchler,
Norbert Dillier, Wouter Dreschler, Matthias Froehlich, Giso Grimm,
Niklas Harlander, Volker Hohmann, Rolf Houben, Sofie Jansen,
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Henning Puder, Michael Schulte, Ann Spriet, Matthias Vormann

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Evaluation of signal enhancement algorithms in the HearCom project

1. Physical evaluation (i.e. through simulation experiments) of a number of state-of-the-art/novel speech enhancement algorithms for future hearing aids using
 - audio files that have been recorded in several real-life acoustic environments
 - performance measures that incorporate various aspects of (impaired) human hearing
2. Selection of 5 promising signal enhancement schemes
3. Real-time implementation on a common hardware/software platform
4. Perceptual evaluation with normal-hearing and hearing-impaired subjects in conditions comparable to real-life listening environments

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Signal enhancement algorithms

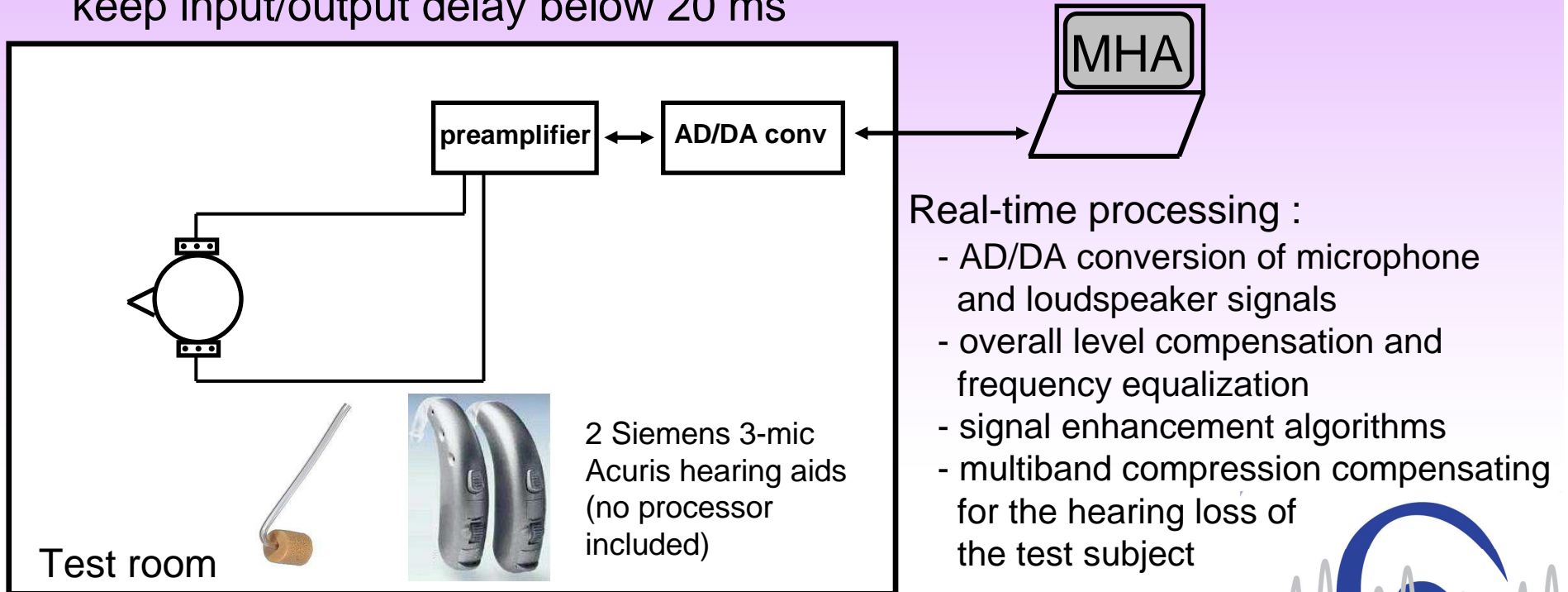
- **SC1:** Single-channel noise suppression based on perceptually optimized spectral subtraction (Samuelsson, 2006)
- **SC2:** Wiener-filter-based single-channel noise suppression (Mauler, 2006; Martin, 2001)
- **BSS:** Broadband blind source separation based on second-order statistics (Aichner et al, 2006; Buchner et al, 2005)
- **MWF:** Spatially preprocessed speech-distortion-weighted multi-channel Wiener filtering (Spriet et al, 2004, 2005; Doclo et al, 2005)
- **COH:** Binaural coherence dereverberation filter (Grimm et al, 2008)

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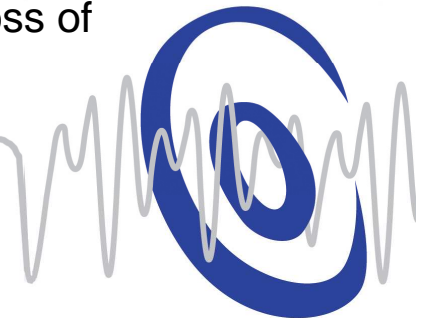


Common evaluation setup

- All algorithms have been implemented in C(++) on a common real-time hardware/software platform called **Master Hearing Aid** (Grimm et al, 2008)
- Laptop PC running low-latency Linux kernel using the Jack sound driver to keep input/output delay below 20 ms



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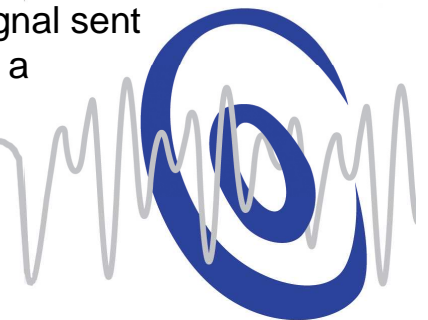
Algorithm comparison

Algorithm	Number of mics used	% CPU time required ¹	Input/output delay ²
SC1	2 x 1 (double monaural)	8.3%	10.8 ms
SC2	2 x 1 (double monaural)	4.2%	16.8 ms
BSS	2 (binaural)	59.9%	10.8 ms
MWF	2 x 3 (double monaural)	4.3%	13.2 ms
COH	2 (binaural)	1.2%	10.6 ms

¹Estimate of the computational complexity of each algorithm measured on a Dell Latitude D610 (Intel Pentium M 1.6 GHz processor) running low-latency Linux. The baseline processing with all signal enhancement algorithms switched off requires 10.3% of CPU time.

²The total input/output delay (MHA + selected signal enhancement algorithm) from the signal sent into the AD converter to the signal that appears at the DA converter output, measured on a Dell Latitude D620 (Intel Core Duo 1.83GHz) running low-latency Linux.

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Perceptual evaluation: test sites and subject groups

- Five test sites, two languages:
 - German:
 - CH-UHZ: Dept. of ORL, University Hospital Zürich, Switzerland
 - DE-HZO: Hörzentrum Oldenburg GmbH, Oldenburg, Germany
 - DE-SAT: Siemens Audiologische Technik GmbH, Erlangen, Germany
 - Dutch:
 - BE-LEU: ExpORL, Katholieke Universiteit Leuven, Belgium
 - NL-AMC: KNO-Audiologie, Academic Medical Center Amsterdam, The Netherlands
- 3 subject groups:
 - Normal-hearing subjects
 - Hearing-impaired subjects: moderate sloping hearing loss
 - Hearing-impaired subjects: moderate flat hearing loss

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Noise scenarios and test rooms

- Noise:
 - Front loudspeaker reproduces speech
 - Left, right and rear loudspeaker output multitalker babble noise
 - The overall noise level is fixed at 65 dB(A)
 - 3 noise scenarios have been considered:
 - in quiet
 - 1 noise source at 90° → point-source noise scenario
 - 3 noise sources at 90°/180°/270° → (pseudo-)diffuse noise scenario
- Two types of test rooms:
 - Room with reverberation time between 0.4 and 0.6 sec, representative for living/office room; is also in compliance with standards for school/educational acoustic environments
 - Reverberating room (reverberation time > 1 sec)

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Adaptive speech reception threshold tests

- Determine the Speech Reception Threshold (SRT) with an adaptive 2-dB step procedure
- The SRT is defined as the average signal-to-noise ratio at which 50% of the presented speech material is intelligible
- Speech material:
 - Dutch: VU sentences
 - German: OLSA sentences
- Test and retest

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Test conditions SRT

TEST CONDITION				ALGORITHM					
Room	Class	Signal	Masker	SC1	SC2	BSS	MWF	COH	Unproc
Living Room	Point-source	0°	Babble 90°	-	-	BE-LEU CH-UHZ NL-AMC	BE-LEU CH-UHZ NL-AMC	-	BE-LEU CH-UHZ NL-AMC
	Pseudo-diffuse	0°	Babble 90/180/270°	ALL	ALL	ALL	ALL	ALL	ALL
	Quiet	0°	-	-	-	-	-	BE-LEU DE-HZO	BE-LEU DE-HZO
Reverberating Room	Diffuse	0°	Babble 90/180/270°	BE-LEU DE-HZO	BE-LEU DE-HZO	BE-LEU DE-HZO	BE-LEU DE-HZO	BE-LEU DE-HZO	BE-LEU DE-HZO
	Quiet	0°	-	-	-	-	-	BE-LEU DE-HZO	BE-LEU DE-HZO

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Adaptive speech reception threshold tests

- The results shown on the next slides are based on data obtained from 109 subjects:

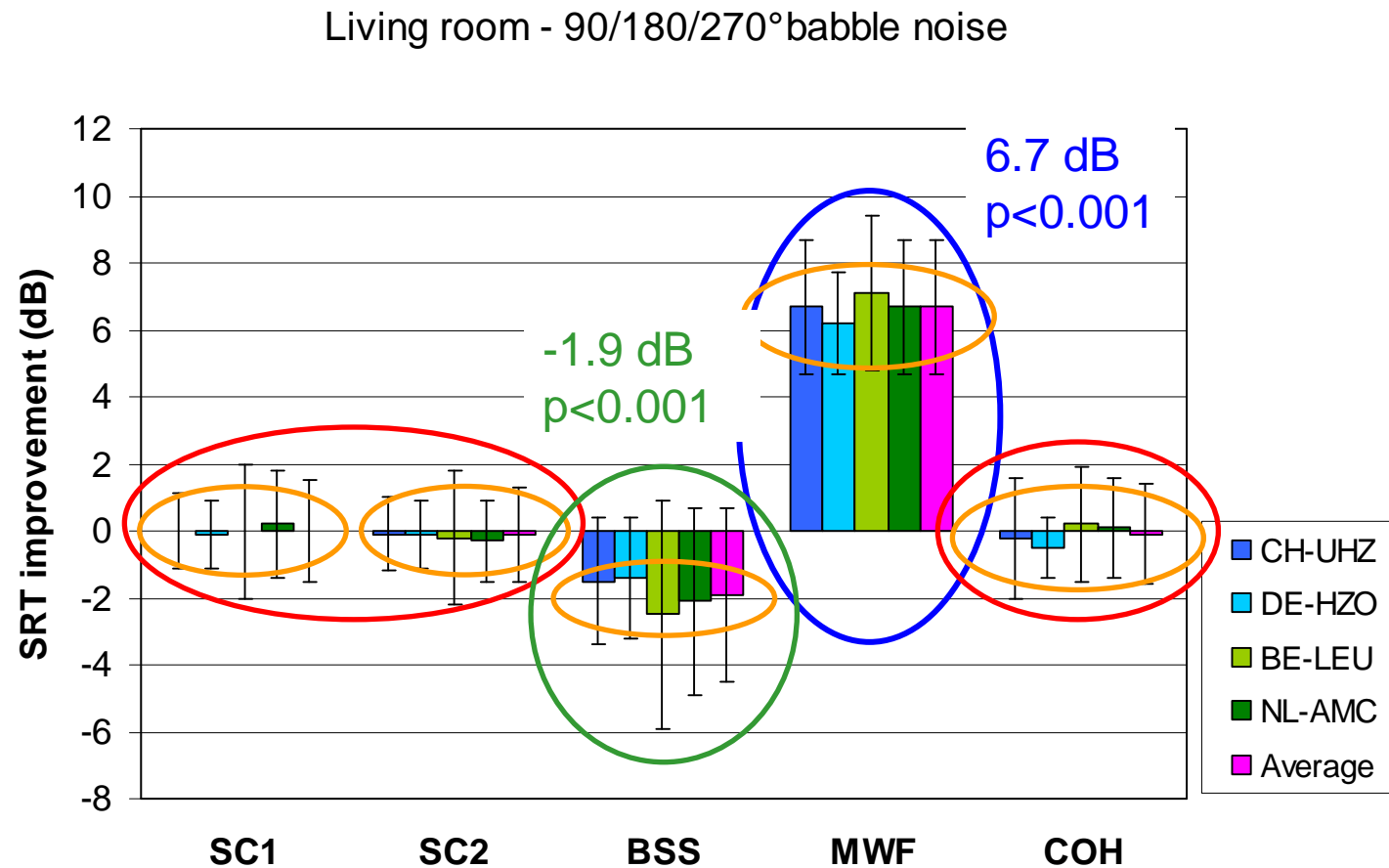
	NH	HI-sloping	HI-flat
CH-UHZ	9	8	4
DE-HZO	9	9	10
BE-LEU	10	10	10
NL-AMC	10	10	10
+ ALL	38	37	34

- The next slides show SRT improvements relative to the unprocessed condition. Hence, positive numbers in the figures indicate an improvement in speech understanding.

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Living room – pseudo-diffuse noise

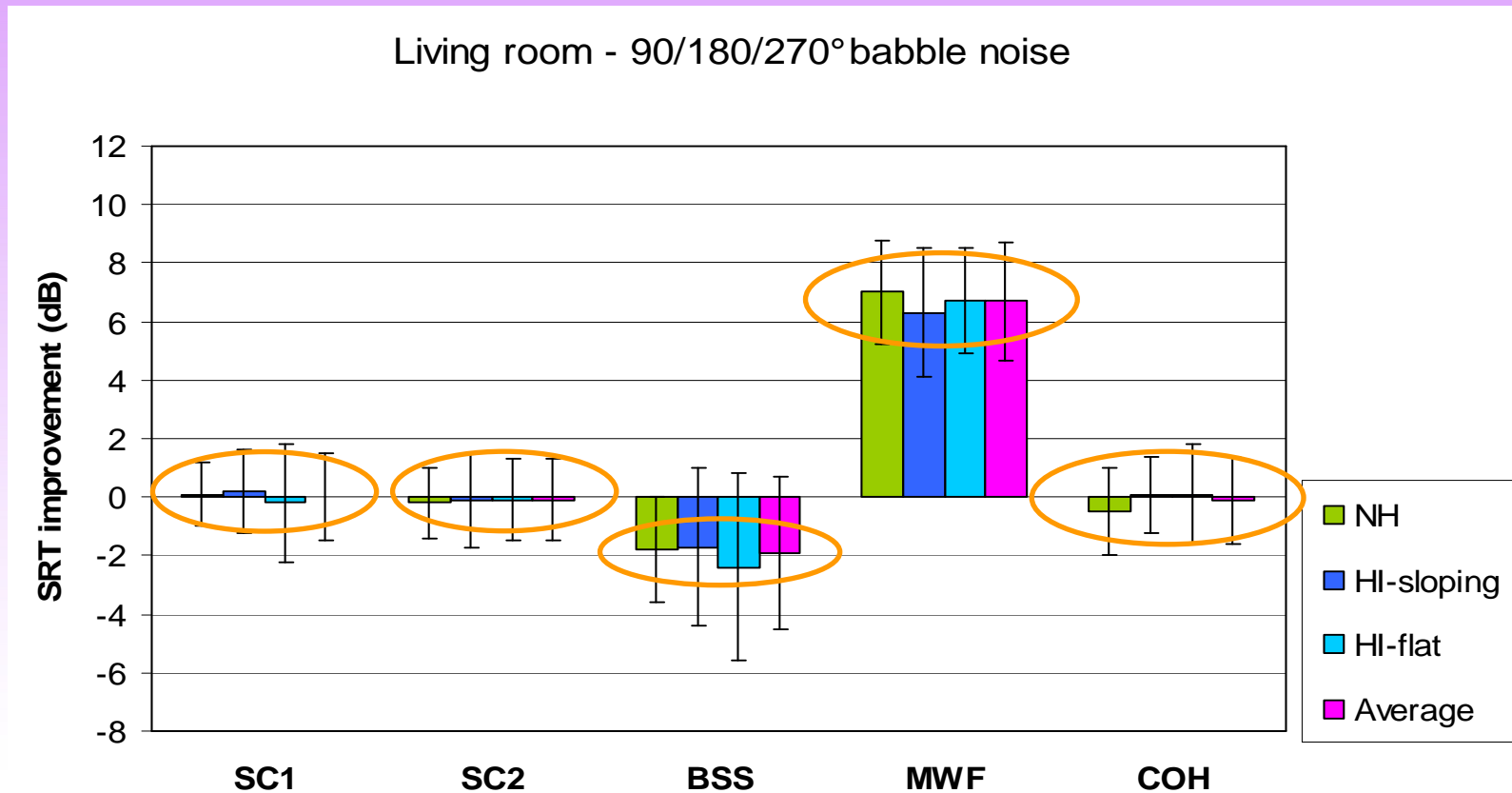


¹SRT improvements relative to the unprocessed condition (i.e. no noise suppression)

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Living room – pseudo-diffuse noise

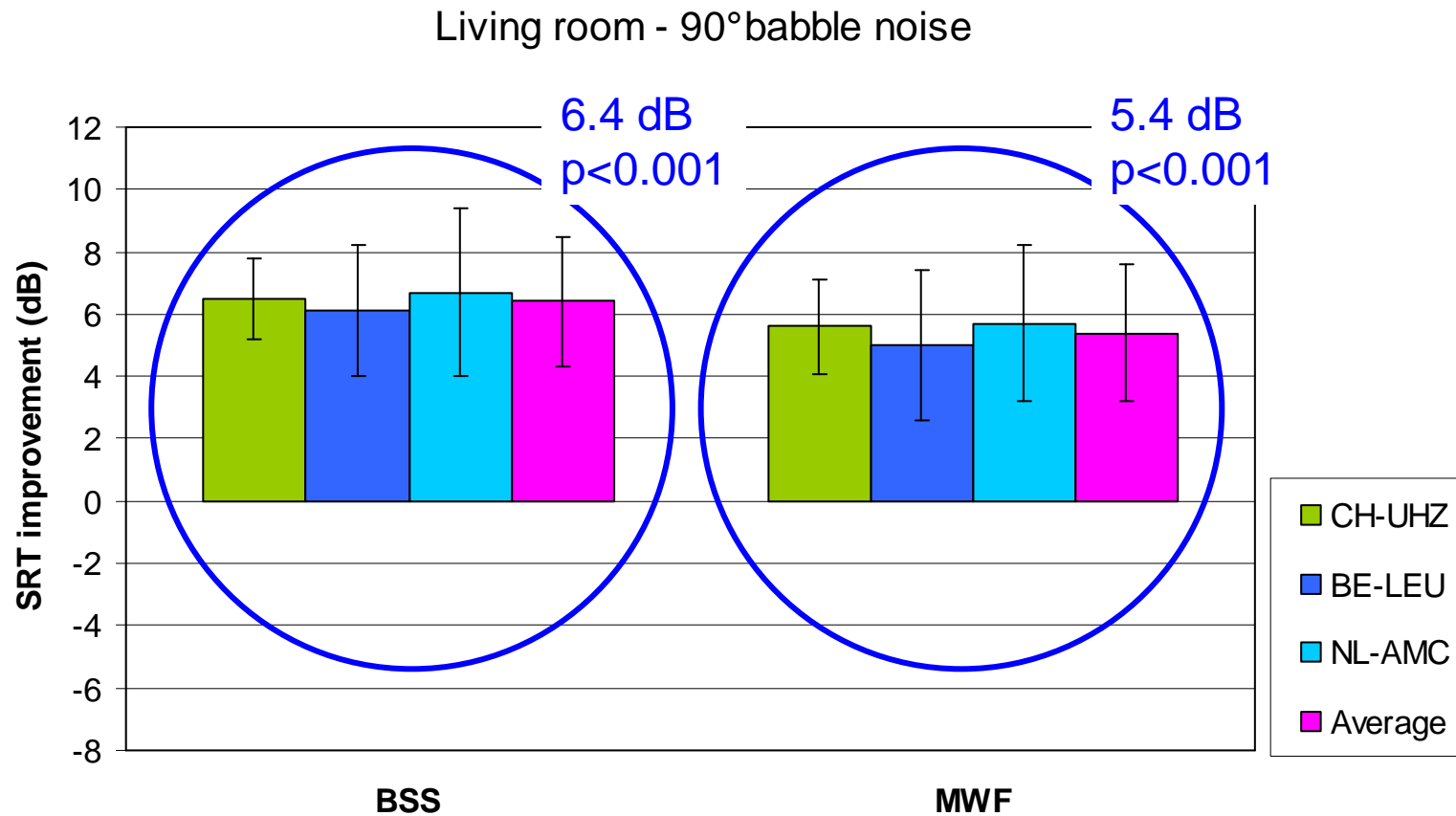


¹SRT improvements relative to the unprocessed condition (i.e. no noise suppression)

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Living room – point-source noise

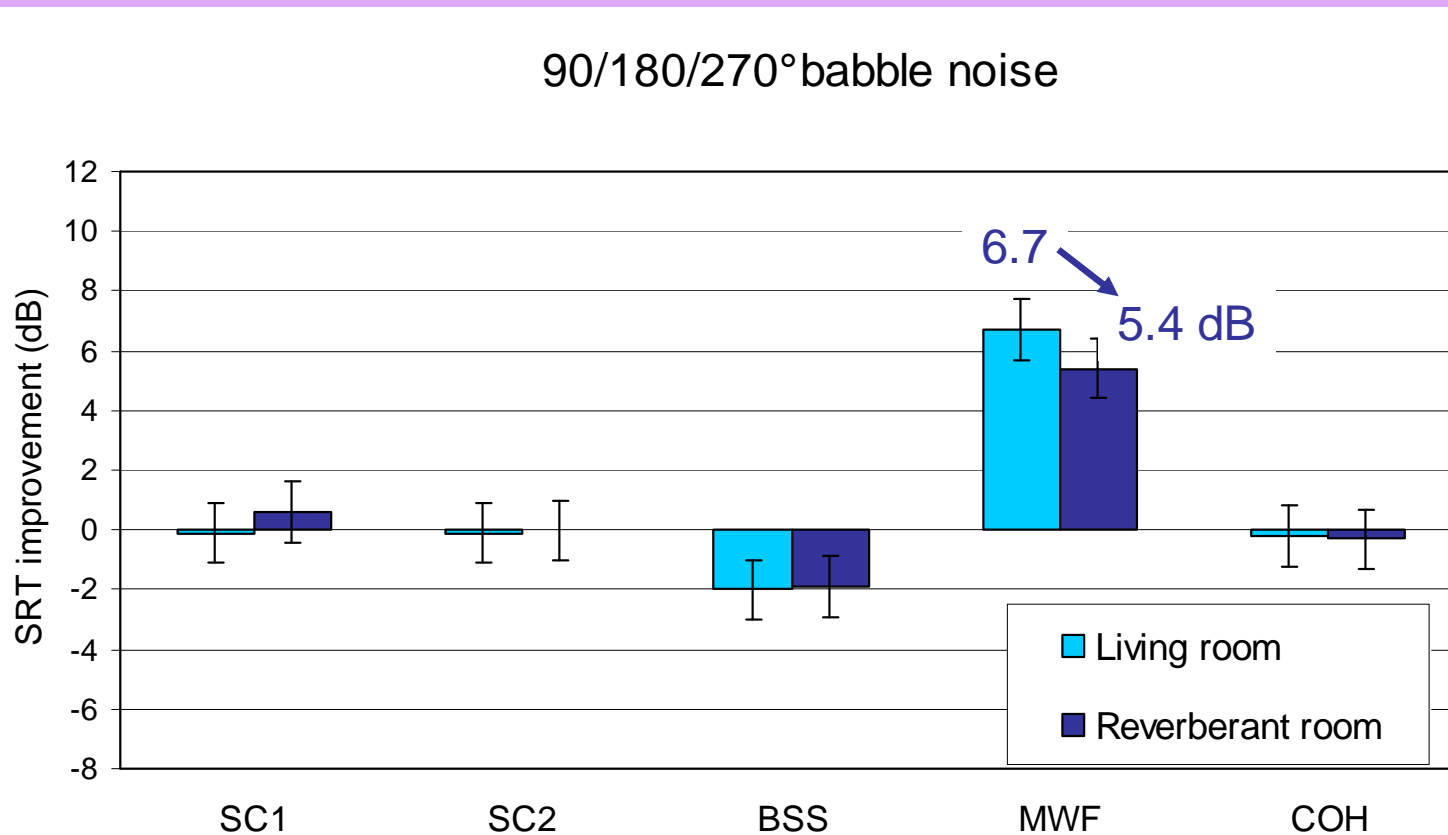


¹SRT improvements relative to the unprocessed condition (i.e. no noise suppression)

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Comparison of test rooms



Remark the absolute SRT-shift of 7 dB for BE-LEU and 5 dB for DE-HZO

¹SRT improvements relative to the unprocessed condition (i.e. no noise suppression)

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Listening Effort Scaling

- Each algorithm and the unprocessed condition is tested
- Setup similar to basic speech test (living room, S0N90/180/270)
- Fixed noise level: 65 dB(A)
- Listening effort scaling is done at 5 fixed SNRs: 0, ± 5 and ± 10 dB
- In total 60 ratings (6 algorithms x 5 SNRs x (test+retest))

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Listening Effort Scaling

Listening Effort Scaling 23

How much effort does it require to listen to and understand the sentences?

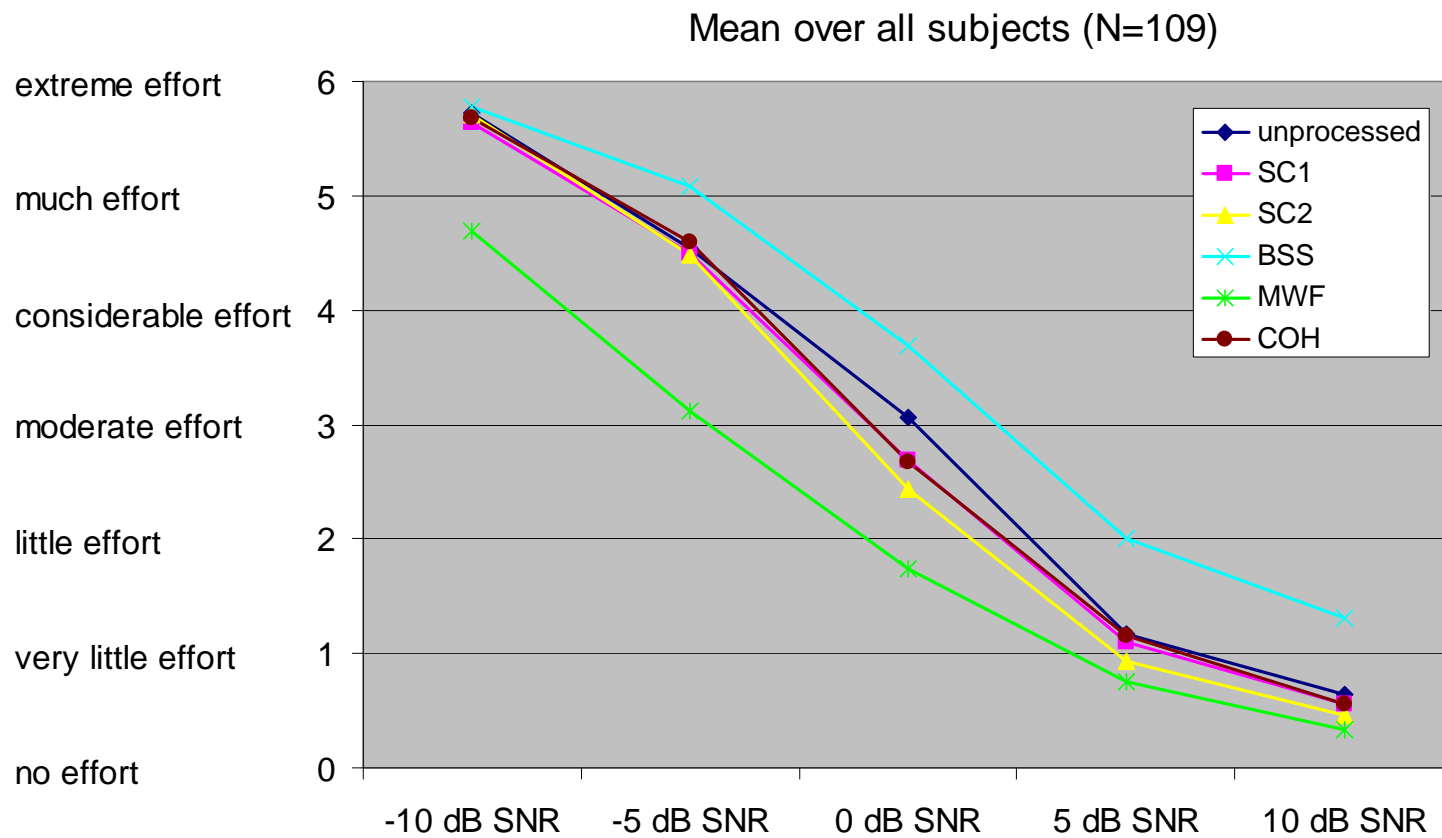
Extreme effort
Much effort
Considerable effort
Moderate effort
Little effort
Very little effort
No effort

Stimulus 1 / 60

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Results LES – living room 90-180-270°



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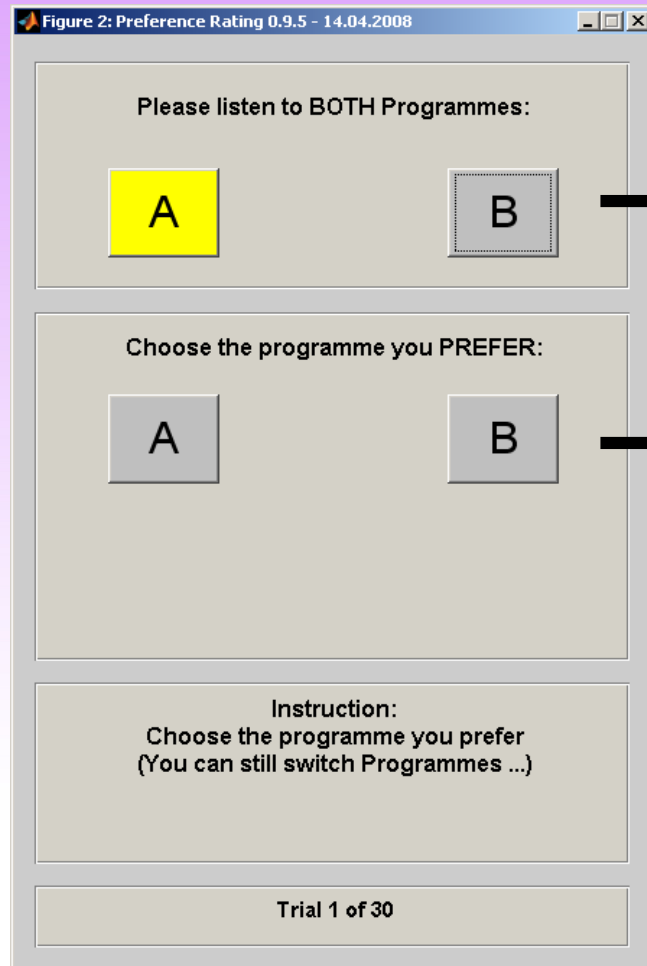
Preference rating

- Each algorithm is tested against the unprocessed condition
- Setup similar to basic speech test (living room, S0N90/180/270)
- Fixed noise level: 65 dB(A)
- Preference rating is done at 3 fixed SNRs: 0, +5 and +10 dB
- In total 30 ratings (5 algorithms x 3 SNRs x (test+retest))
- Instruction: 'Please imagine that you are sitting in a noisy room and trying to follow a conversation, while other persons are talking.
Which of the programs would you like most to have in your hearing aid in this situation?'

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Preference rating



→ 1. LISTEN

→ 2. CHOOSE

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Preference rating

Figure 2: Preference Rating 0.9.5 - 14.04.2008

Please listen to BOTH Programmes:

A **B**

Grade your preference:

A very much better than B

A much better than B

A better than B

A slightly better than B

A very slightly better than B

Instruction:
Please grade your preference
(You can still switch Programmes...)

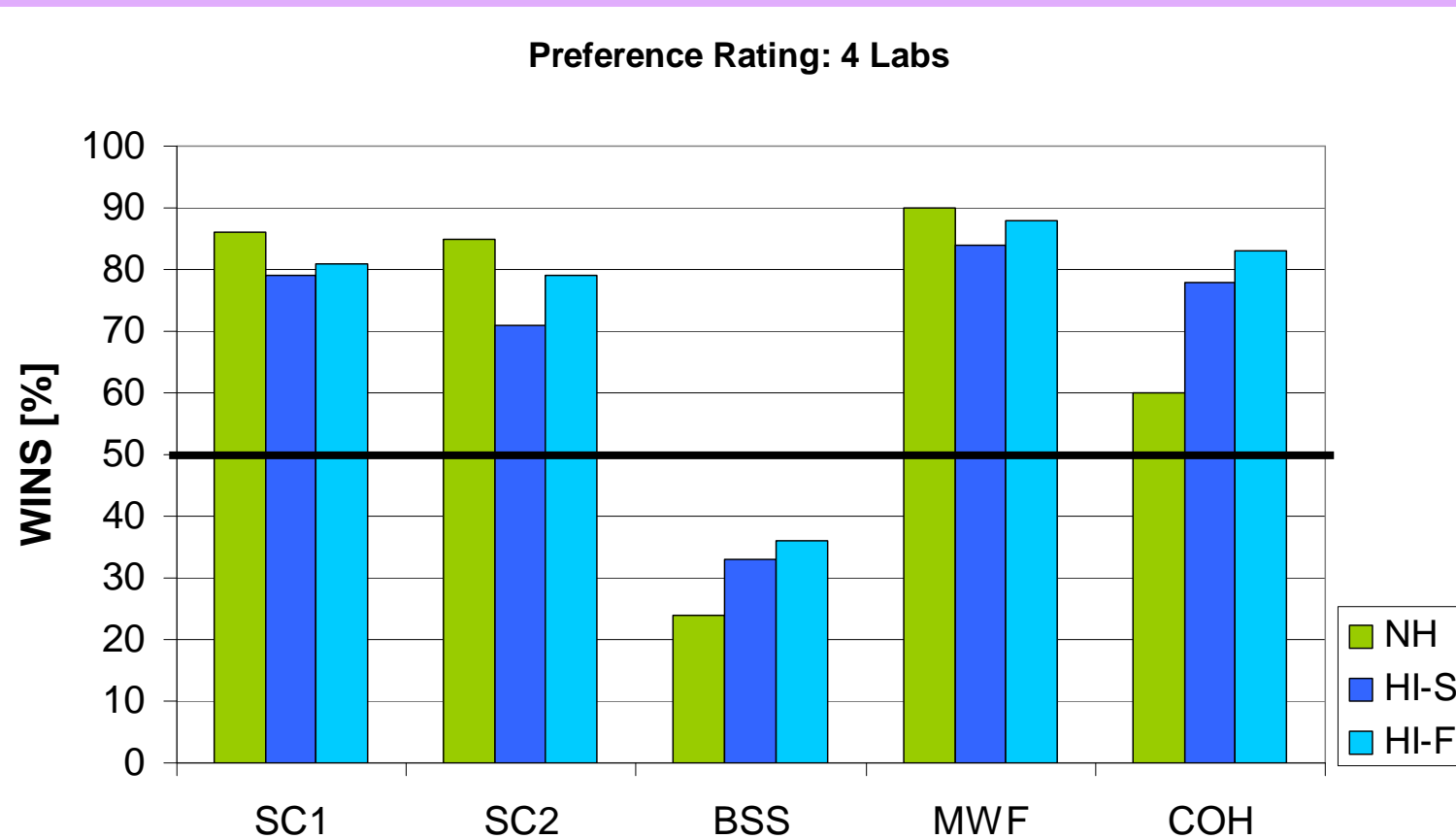
Trial 1 of 30

→ **3. GRADE**

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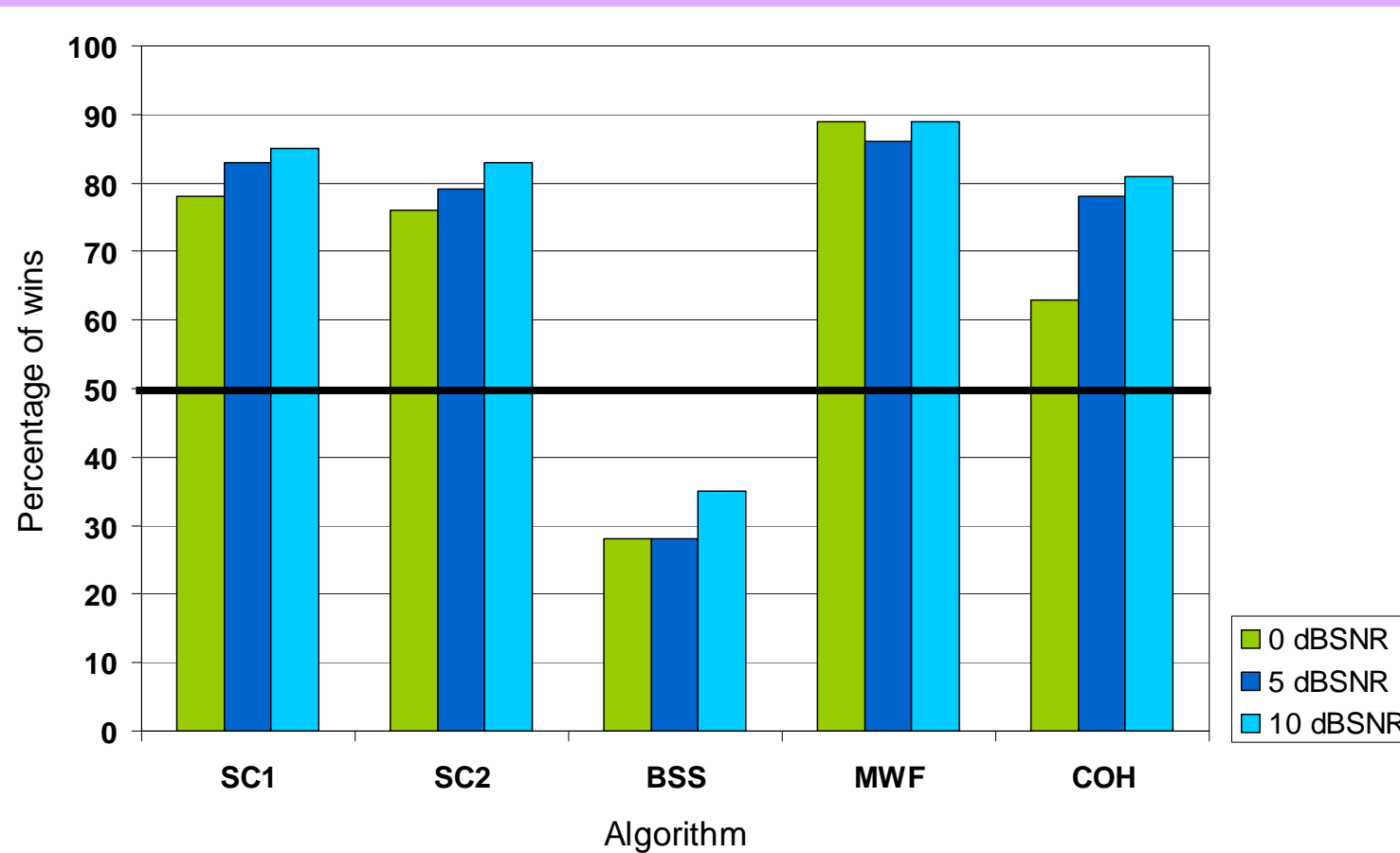
Results PR – living room 90-180-270°



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Results PR – living room 90-180-270°



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Conclusions (1)

SRT tests

Improvement relative to the unprocessed condition:

- Good test-retest reliability within test sites and good correspondence between test sites and between languages!
 - test protocol is important, but exact test conditions are not crucial
- After correction for frequency-dependent hearing loss: no differences between subject groups!
 - tests in normal-hearing subjects can predict performance in hearing-impaired subjects.

In living room conditions – 3 noise sources:

- Only with MWF an SRT improvement relative to unprocessed is obtained.

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Conclusions (2)

Listening Effort Scaling

- Results depend on SNR
- Results show comparable trends as the SRT results

Preference Rating

Although algorithms other than MWF do not provide improvements in SRT, subjects do prefer them over the unprocessed condition

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Physical evaluation

- To properly validate a speech enhancement algorithm advanced physical evaluation measures are required that incorporate various aspects of (impaired) human hearing, and that can reliably predict algorithm performance through simulation experiments only:
 - Speech intelligibility index (SII)
 - Segmental SII (segSII)
 - Signal-to-noise loudness level difference (SNLL)
 - Signal excitation-level distortion (SED): measure of spectral deviation between processed and unprocessed target signals
- We relied on multi-microphone audio files that have been recorded with a hearing aid dummy in several real-life acoustic environments:
 - Speech in low-reverberant room and living room with interfering speech, speech shaped noise, or music at different angles of incidence
 - Speech in cafeteria noise, car noise, street noise
 - As speech and noise files were recorded separately a scenario with whatever SNR wanted can be reconstructed

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MHA interface

The screenshot displays the MHA control interface. The main window, titled "mhacontrol - Control panel for the HoerTech MHA", features a "PHS control" header with logos for HearCom, Carl-Neuberg Universität Oldenburg, and HörTech. The interface is divided into several sections:

- Algorithm:** A list of algorithms including (none), identity, SE_KTH_sc, DE_RUB_sc, DE_UEN_bss, BE_LEU_beam, and DE_UOL_coh.
- Actions:** Buttons for Fitting, in-situ signals, and Levels.
- Administration:** Buttons for Audio connections, Calibration, and Advanced control.
- Status:** Displays Calibration ID (2007-09-18 acuris curtain room), Client ID (_P000101), Gain prescription rule (NALRP), and Preset (First fit, NALRP, 2007-09-18).
- Quit:** A red 'X' button.

An "MHA Calibration" window is overlaid, showing a list of calibration events for "Acuris P + Multiface I (-10 dBV) PRELIMINARY real ear default (flat, 0 dB FS = 120 dB SPL)":

- 2007-09-12
- 2007-09-13
- 2007-09-13 acuris
- 2007-09-17 acuris
- 2007-09-17 acuris galmkamer
- 2007-09-18 acuris curtain room

The window includes "New", "Remove", and "Close" buttons. Below the list, a table shows calibration results:

left_front	119.4	left	143.8
left_mid	119.2	right	144.5
left_rear	118.9		
right_front	119.7		
right_mid	120.2		
right_rear	120.0		

A graph at the bottom right of the calibration window shows a red line representing a frequency response curve, with frequency on the x-axis (250 to 8000 Hz) and amplitude on the y-axis (-20 to 20 dB).

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Selected signal enhancement algorithms (1)

SC1 : Single-channel noise suppression based on perceptually optimized spectral subtraction (Samuelsson, 2006)

- Low-delay single-channel noise suppression algorithm based on spectral subtraction
- By carefully selecting the amount of under- or oversubtraction, the enhanced signal is perceptually optimized to eliminate musical noise artifacts
- As SC1 is a single-channel noise suppression algorithm, it cannot take advantage of the spatial diversity in the test setup

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Selected signal enhancement algorithms (2)

SC2 : Wiener-filter-based single-channel noise suppression (Mauler,2006; Martin, 2001)

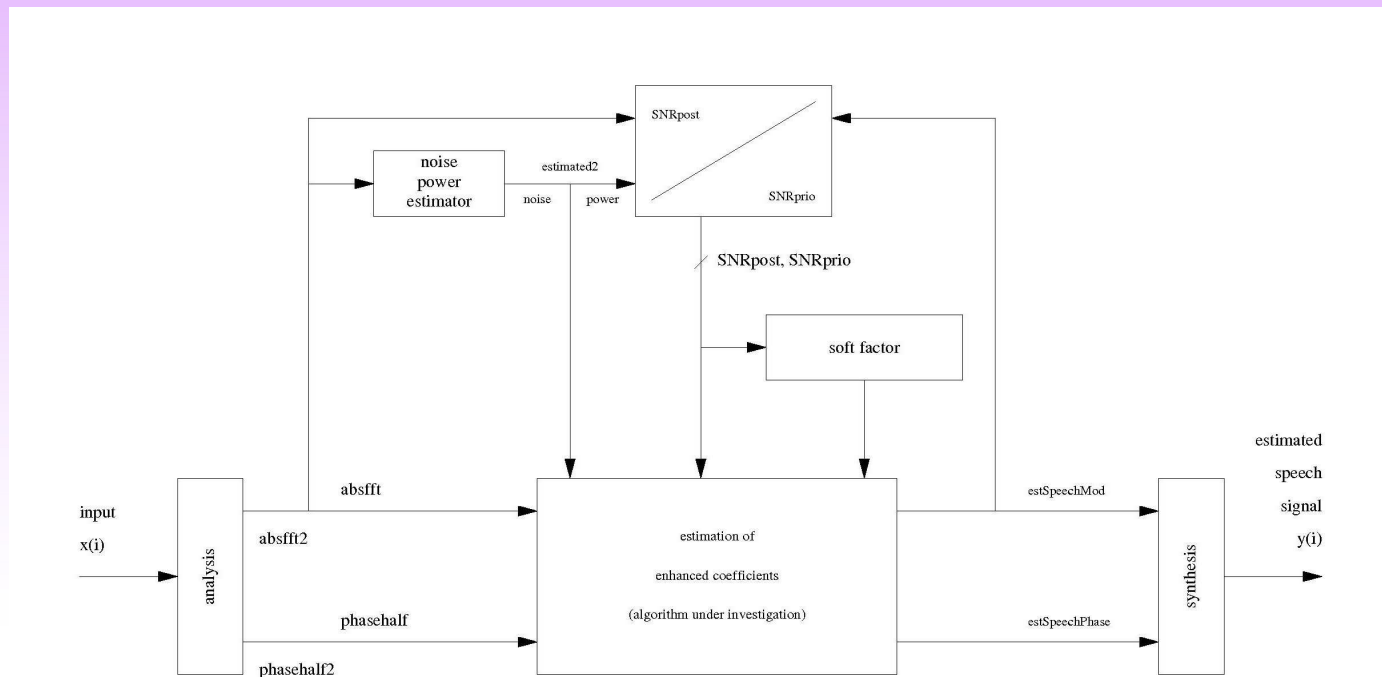
- Low-complexity, low-delay single-channel noise suppression algorithm relying on Wiener filter-based minimization of the mean squared error between the (unknown) desired speech signal and a filtered version of the observed noisy speech
- As SC2 is a single-channel noise suppression algorithm, it cannot take advantage of the spatial diversity in the test setup

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Selected signal enhancement algorithms (2)

SC2 : Wiener-filter-based single-channel noise suppression
(Mauler,2006; Martin, 2001)



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Selected signal enhancement algorithms (3)

BSS : Broadband blind source separation based on second-order statistics
(Aichner et al, 2006; Buchner et al, 2005)

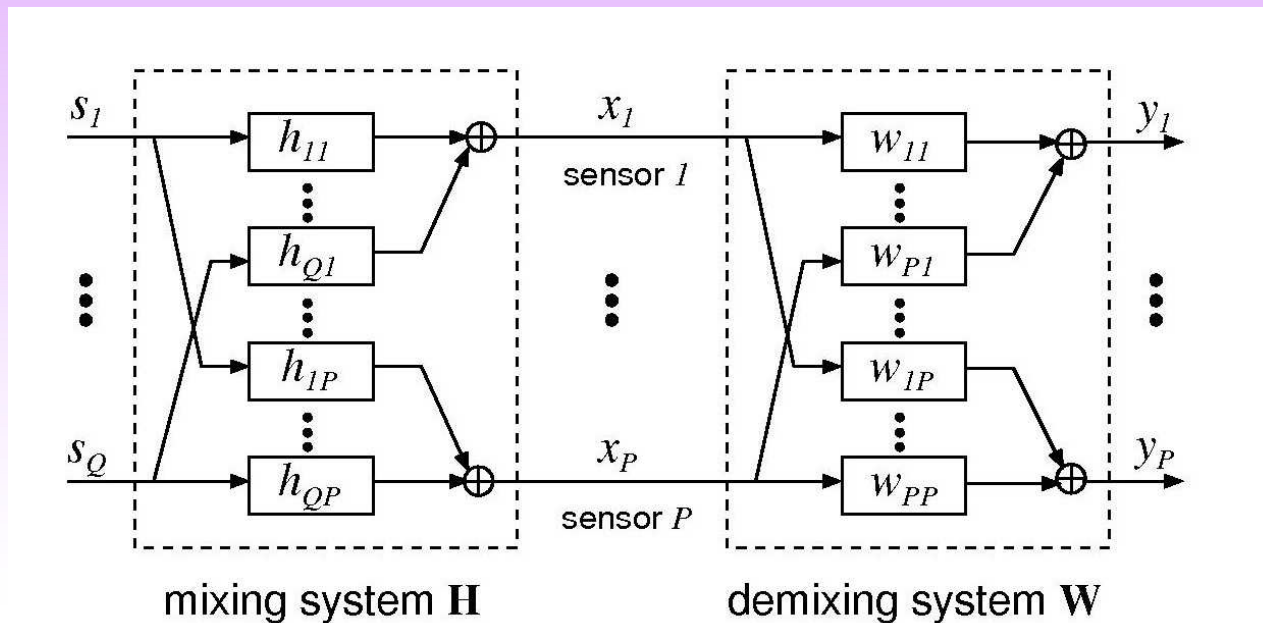
- Low-complexity, low-delay broadband frequency-domain blind source separation algorithm based on second-order statistics
- No geometric information about the placement of the sensors is needed, but it is assumed that the desired source is located in front of the hearing aid user
- In the HearCom project a 2-channel variant of the algorithm has been considered, which is able to separate two point sources

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Selected signal enhancement algorithms (3)

BSS : Broadband blind source separation based on second-order statistics
(Aichner et al, 2006; Buchner et al, 2005)



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Selected signal enhancement algorithms (4)

MWF : Spatially preprocessed speech-distortion-weighted multi-channel Wiener filtering (Spriet et al, 2004,2005; Doclo et al, 2005)

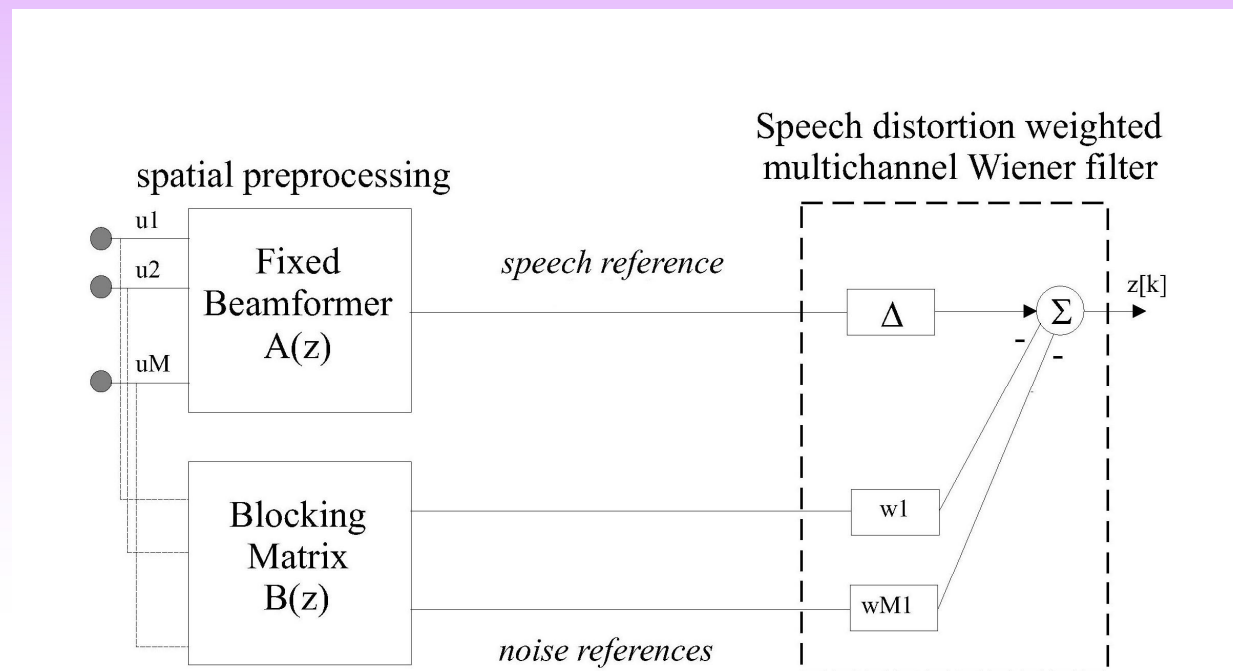
- Spatially preprocessed speech-distortion-weighted multi-channel Wiener filtering based adaptive beamforming algorithm
- Can be viewed as a variant of the generalized sidelobe canceller (GSC) structure
- Trades off between noise suppression and speech distortion
- In the HearCom project we considered a 3-channel variant of the algorithm

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Selected signal enhancement algorithms (4)

MWF : Spatially preprocessed speech-distortion-weighted multi-channel Wiener filtering (Spriet et al, 2004,2005; Doclo et al, 2005)



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Selected signal enhancement algorithms (5)

COH : Binaural coherence dereverberation algorithm (Grimm et al, 2008)

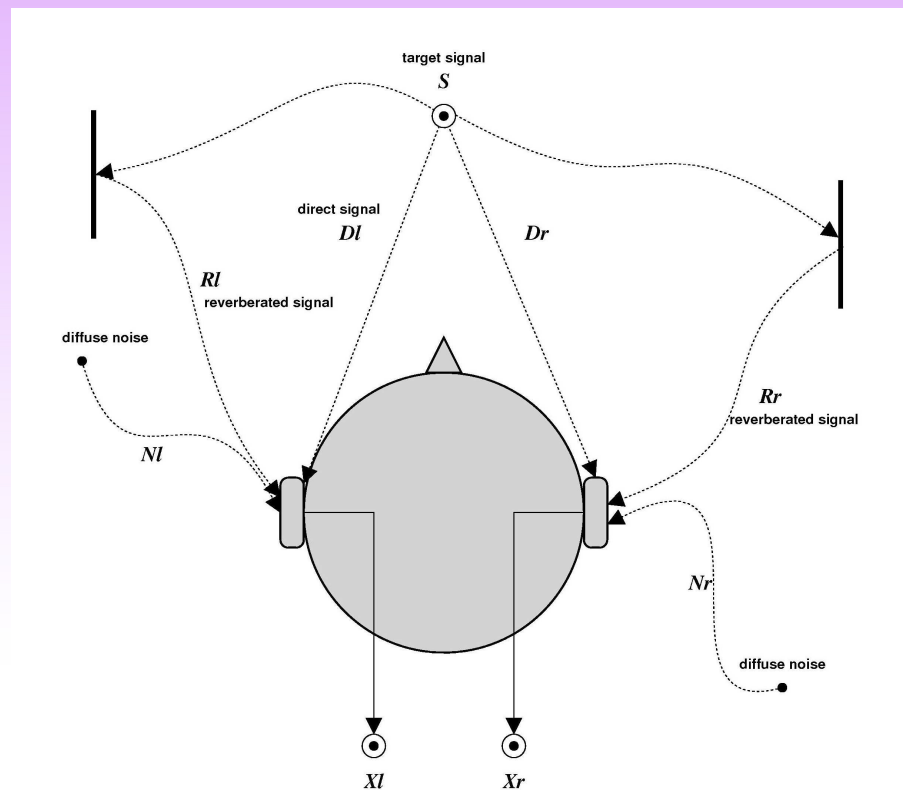
- A binaural coherence filtering based approach designed to increase listening comfort and speech intelligibility in reverberating environments and diffuse background noise (e.g. babble noise)
- It estimates the coherence between the signals captured at the left and the right ear. The estimate is computed in different frequency bands using an FFT-based filterbank with a non-linear frequency mapping that approximates a Bark scale
- COH is a binaural algorithm producing stereo output

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Selected signal enhancement algorithms (5)

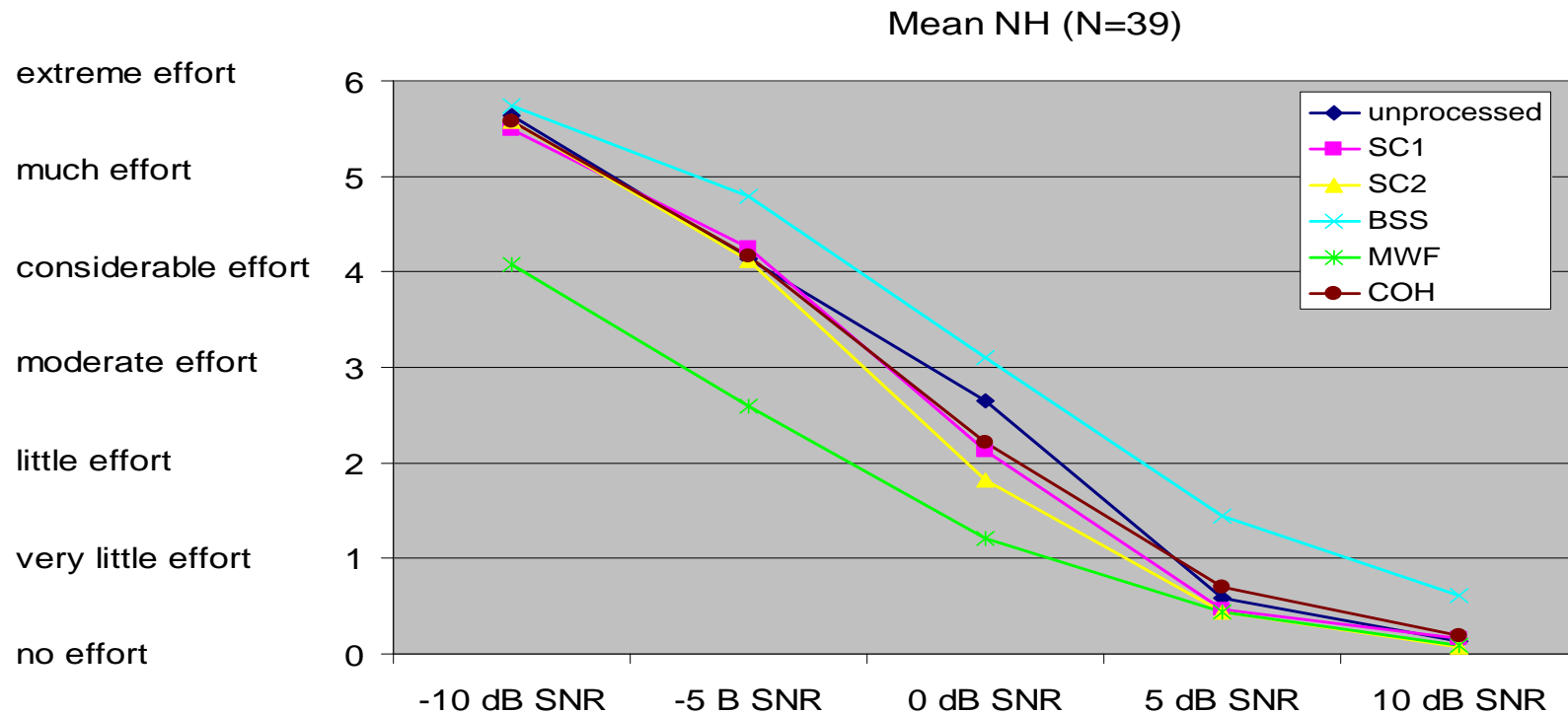
COH : Binaural coherence dereverberation algorithm (Grimm et al, 2008)



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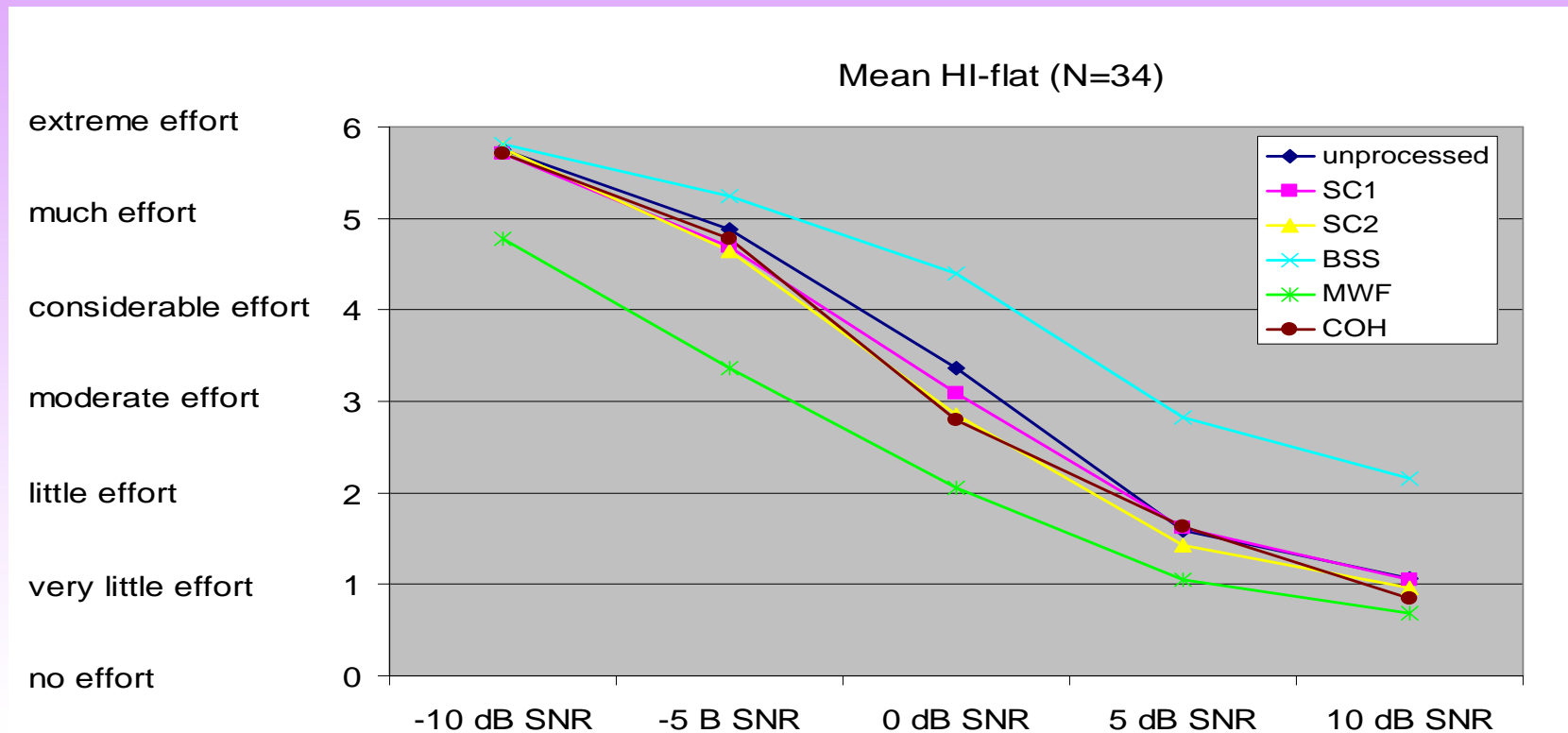
LES NH



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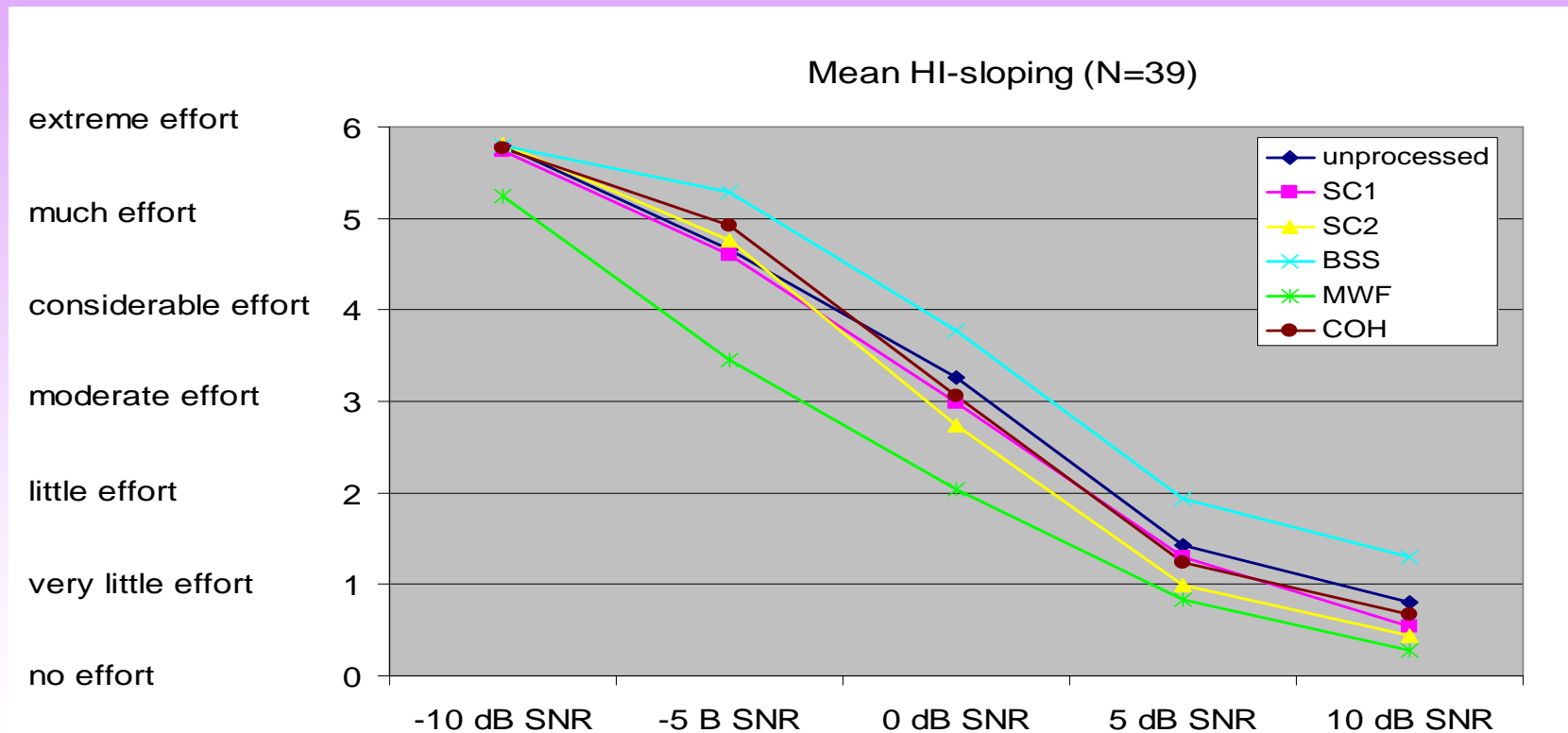
LES HI-flat



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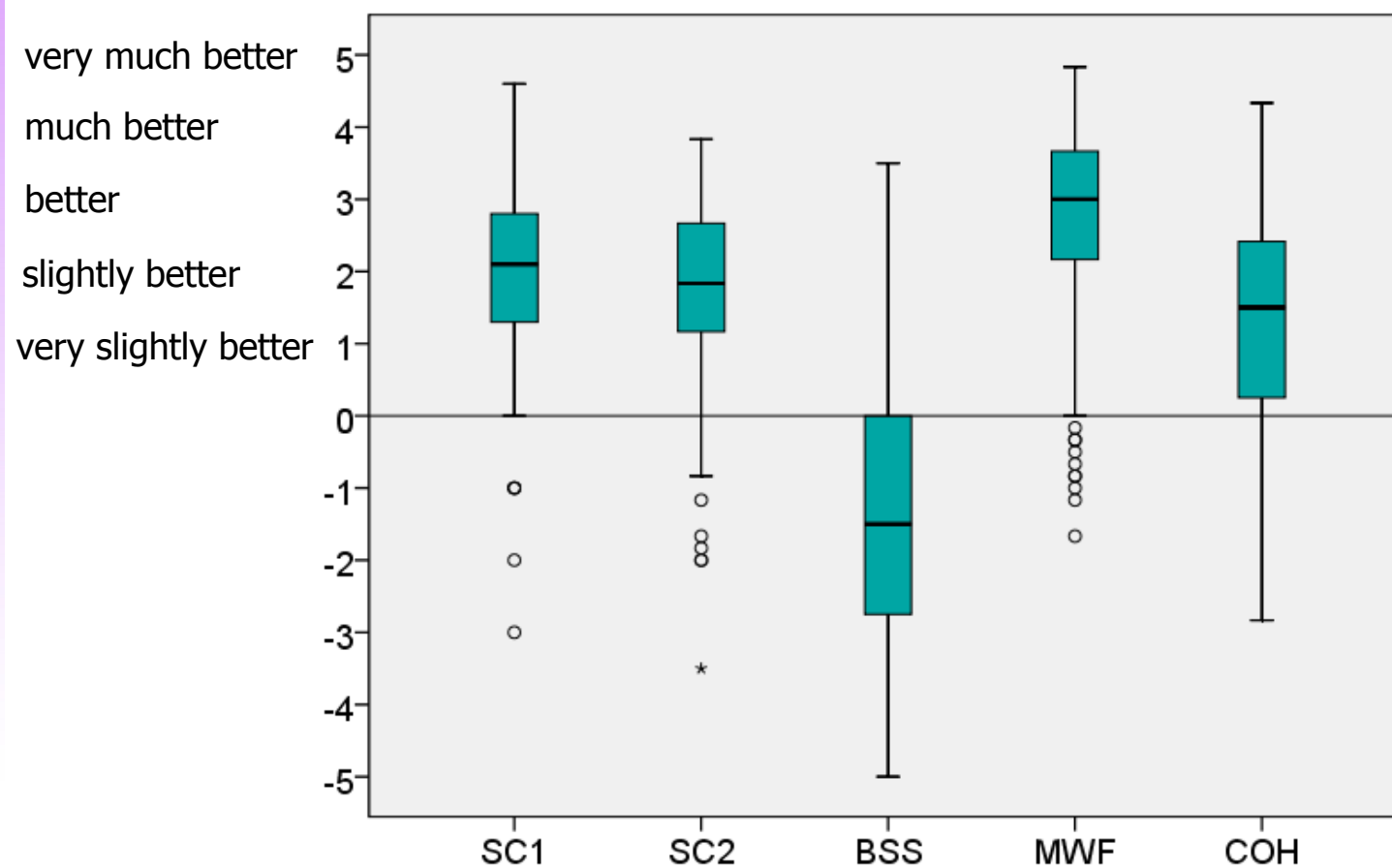
LES HI-sloping



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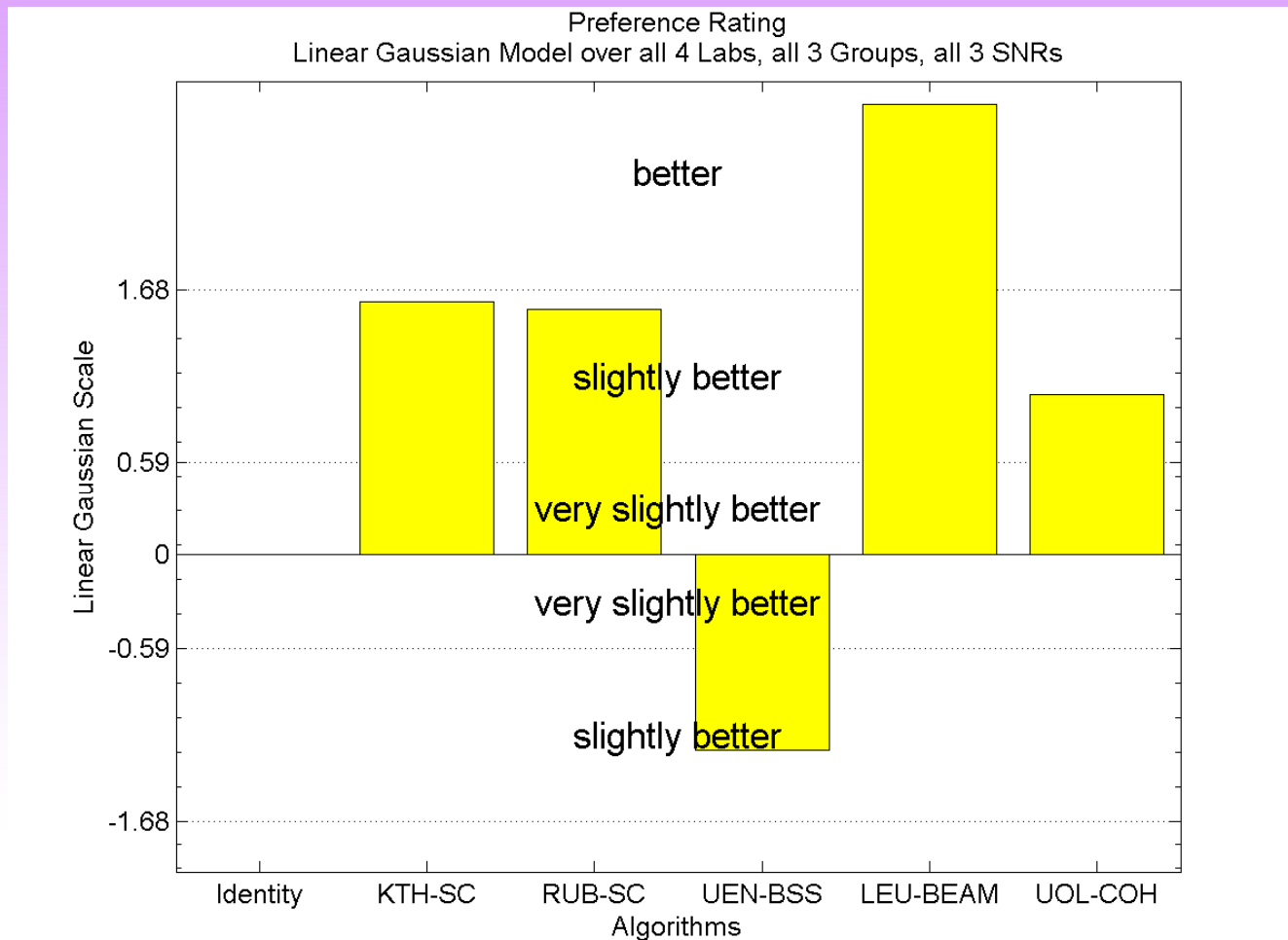
Results PR – living room 90-180-270°



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Results PR – living room 90-180-270°



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