EBU – TECH 3310

Possible use of Spectral Band Replication in DAB

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Possible use of Spectral Band Replication in DAB

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Executive Summary

Background

The present report was produced by Project Group B/AIM (Audio in Multimedia) in the study period 2004/2005. The report was requested by the EBU Broadcast Management Committee (BMC) in order to assess whether or not it would be worthwhile to introduce Spectral Band Replication (SBR) in the existing Digital Audio Broadcasting (DAB) system[1].

The DAB system uses a relatively old audio coding technology, MPEG Layer 2. As DAB uses Unequal Error Protection (UEP), it would be very difficult to replace MPEG Layer 2 with a radically new coding system which may be more efficient but would disenfranchise the current receivers. To this end, EBU launched a study which was intended to show whether or not SBR could make DAB a more spectrum-efficient broadcasting system, however without significantly penalizing the existing DAB receivers that have been already rolled out in the consumer market.

SBR is a relatively new procedure which has been designed to improve subjective audio quality while keeping the overall bit rate low. SBR was also intended to allow for backwards compatibility (i.e. reuse of existing receiver infrastructure, while a broadcaster would upgrade the transmissions to accommodate SBR). SBR might be most effective in a "green field" situation (in which no legacy problems exist).

SBR has been successfully applied to Advanced Audio Coding (AAC) and other advanced audio coding technologies, which make use of the perception models. For example, SBR is being used with AAC and is part of the ISO/IEC standard (14496-3, incl. Amd.1:2003, Amd.2:2004, and all corrigenda). The AAC, enhanced by SBR, is called aacPlus and is widely standardized by many international standardization bodies. aacPlus v2 is specified as the high quality audio codec in 3GPP (3rd Generation Partnership Project), and all of its components are part of MPEG-4. aacPlus v1 is standardized by 3GPP2, Internet Streaming Media Alliance (ISMA), Digital Video Broadcasting (DVB), the DVD Forum, Digital Radio Mondiale (DRM), and many others.

The SBR technology has been developed by the Coding Technologies. It is using the correlation between low and high frequency spectral components at harmonic audio signals like music and speech in the way of replicating the spectral portion of the lower bands into the higher one and only the envelope of the higher spectral bands is necessary in transmission for detecting the higher spectral portions at the receiver side. The saved bits can be used for an much more effective coding in the lower bands especially when coded with half-sampling-rate, making the doubled number of lower spectral bands available (Dual Rate Mode at SBR).

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[1] EN 300 401: Digital Audio roadcasting (DAB) to mobile, portable and fixed receivers, ETSI, Version 1.3.3, May 2001

Project Group B/AIM, under the chairmanship of Gerhard Stoll (IRT), launched extensive studies in order to determine the suitability of SBR for DAB. The Group divided the work into two parts, as follows:

Part 1: This part is titled ***Error sensitivity***. The purpose of this part is to assess to what extent SBR affects the error performance of the DAB system. In particular, two critical system parameters were studied:

- **Onset of Impairments (OoL)**; this is the point of carrier-to-noise ratio (C/N) at which first subjectively noticeable errors appear
- **Point of Failure (PoF)**; this is the point at which the system breaks down (i.e. a total services failure)

Part 2: This part is entitled: ***Subjective Quality Evaluations of SBR***. The purpose of this part was two-fold, as follows:

- to assess coding gain due to SBR, compared to conventional MPEG Layer 2, and
- to assess loss of quality in existing DAB receivers (i.e. compatibility).

Principal conclusions of Part 1 of Tech 3310

Part 1 (Error Sensitivity) shows that the impact of SBR coding on the radio-frequency (RF) system performance is low:

At high audio bitrates (192 kbit/s) SBR improves C/N by 1.0-1.5 dB for "rural" and 0.5-1.0 dB for "urban" model.

At low bit rates (e.g. 48 kbit/s) SBR causes C/N degradation of 0.75 dB for "rural" and 0.4 dB for "urban" model.

Our study shows that the coverage area would not change significantly if SBR is introduced in DAB. In other words, the RF performance in terms of C/N will not be significantly improved or degraded if SBR is introduced to DAB. The existing and new receivers could be used in the same area. From this point of view, the backward compatibility criterion could be fulfilled.

Principal conclusions of Part 2 of Tech 3310

Part 2 (Subjective Quality Evaluations of SBR) considers how much coding gain could be achieved in the DAB system if Spectral Band Replication is introduced *in both the broadcast signal and in the DAB receivers*.

Our study has shown that SBR can achieve some coding gain. This means that the same subjective audio quality can be achieved at a lower bitrate (compared to audio quality of MPEG Layer 2).

For example, the quality of enhanced DAB system using SBR at 64 kbit/s is equivalent to the conventional DAB system (which uses "old" MPEG Layer 2 audio coding) at 96 kbit/s. The coding gain in this case amounts to 33%, which is quite significant, particularly in areas where competition for spectrum is high. As the bitrate increases above 100 kbit/s, coding gain diminishes. Indeed, at 160 kbit/s and above the coding gain is close to zero. This can be explained quite easily: the intrinsic quality of conventional MPEG Layer 2 is of such a good quality already that SBR cannot improve it any further.

Our study shows that SBR is most efficient at lower bitrates (with half-sampling rate) and when joint stereo mode are used.

The downside of SBR is that the existing receivers (that do not implement SBR in the decoder) may be penalised. Our study shows that penalty is higher at lower bitrates. For example, at bitrates 128 and 112 kbit/s the penalty becomes quite important and it amounts to about 20 points on the MUSHRA scale. At 96 kbit/s the penalty amounts even to about 40 points ("good" becomes "poor" to "bad").

Overall conclusions

The introduction of SBR in DAB involves several considerations, both technical and non-technical. The present document focuses on the matters involving the technical performance of the SBR inclusion only.

However, there are many other issues that need to be considered before any decision about using SBR in DAB is taken.

One of the issues is the added complexity on the broadcast and receiver end, and the resulting increase in cost of emission and reception. To this, one should add the intellectual property and licensing issues.

A major unknown is the market price increment of DAB receivers (which are already relatively expensive compared to the analogue ones). It is uncertain when and how the receiver manufacturers would support the introduction of SBR in their commercial products. The WorldDAB Forum has so far not shown enough enthusiasm to embark into SBR either.

The SBR benefits, particularly those related to a higher spectrum efficiency, should be balanced against the potential problems and costs. The analysis shows that, at the moment, SBR could improve technical quality of the DAB system, particularly if bitrates 64 kbit/s (or lower) and joint stereo mode are used. This advantage is to be counterbalanced with the audio quality degradation of the existing (non-SBR) receivers. This may be in some cases as big as two categories (i.e. 40 points on MUSHRA scale).

To this end, the general conclusion reached by this study is that SBR is not to be recommended in the areas where DAB services are commercially rolled out.

Acknowledgements

The EBU would like to thank all active members of the Project Group B/AIM (Audio In Multimedia), in particularly the Chairman Gerhard Stoll (IRT) for his guidance and Wolfgang Krafft (IRT) who supervised all the laboratory tests and evaluations with enthusiasm and perseverance.

A note about the language used

Because this paper including the pictures originally was written in German language the following translation table is included here:

German	English	Comment
Codiergewinn	coding gain	
Qualitätsgewinn	quality gain	
Cembalo	harpsichord	
Klassik	classical music	
Pop	pop (music)	
Sprache	speech	male
Gesang	solo voice (singer)	female
Stadion	(sports) stadium	ice hockey match
Mittelwerte	average values	
Vertrauensbereich	confidence interval	
über alle getesteten	over all tested	
bei	at	
Bitrate	bit rate	
Audioqualität	audio quality	
als Funktion	as function	
Gewinn	gain	
gewonnene Bitrate	saved bit rate	
extrapoliert mit	extrapolation process with	

Part 1: Error Sensitivity

1 Purpose and Scope of Part 1

The DAB system uses the MPEG audio coding system, known as MPEG-1 Layer 2. This system was standardized in 1993 as ISO/IEC 11172-3. In 1997 the system was extended to embrace the MPEG-2 standard ISO/IEC 13818. The latter standard allows for half-sampling for low audio data rates, and thus improves audio quality at low bit rates - but it uses the same number of spectral bands.

The work of Part 1 has mainly been conducted at IRT [1]. Some tests have also been carried out by NRK [2]. The work has been coordinated by Project Group B/AIM.

The purpose of our tests in Part 1 was to investigate differences in transmission performance between SBR and normal MPEG Layer2 coded audio signals. In order to carry out these tests, a DAB Channel Simulation tests were needed.

In order to assess the error performance of the conventional DAB system using MPEG Layer 2, the BBC Research and Development Department has carried out several subjective tests within the EUREKA 147 project back in 1994 [3] and 1996 [4].

Almost ten years later, we took the BBC tests as a basis for our tests. However, instead of MPEG Layer 2 (L2 Normal), we used SBR-MPEG Layer2 plus.(SBR). As the BBC tests found that similar results were obtained in VHF Band III and L-Band, we only performed out tests in Band III.

2 Test Conditions

For compatible results we chose a test set comparable to that taken by the BBC. However, instead of Grundig Fading Channel Simulators (FADICS) which was used by the BBC but was no longer available, we used the Rohde & Schwarz TV signal generator SFQ-B11 with integrated noise generator and fading simulation options. The latter was calibrated such that for non-SBR conditions gave the same results as FADICS. With this equipment the real transmission performance could be simulated (i.e. Gaussian and Rayleigh conditions in an rural or urban environment).

We used the same audio test signals from the EBU SQAM CD as the BBC:

Three music examples (AB=Abba, CL=clarinet, KL=Glockenspiel) and
two speech examples (FR=female voice, MA=male voice)

For encoding in SBR Layer2plus or Normal Layer2 we used a software package received from Coding Technologies. This package allowed us to configure some optional parameters such as bitrate, mode etc.

We used a "Scout" DAB receiver. This receiver has an RDI-USB-interface which allows to connect a PC to record the transmitted DAB data stream as „mp2“ file. Unfortunately it was not possible to replay the decoded audio data stream in real time, so we had to record each DAB data stream separately with a different C/N ratio both for Gaussian Channel and for Rayleigh Channel with the configuration “Rural/Urban”. We tested error performance at three different Error Protection Levels: PL1=0.35=high; PL3=0.5=medium; PL3=0.75=low. In the case of Gaussian Channel a high resolution was achieved by C/N steps of 0.5 dB. For Rayleigh Channel (Rural/Urban) an increment of 1.0 dB was used. The C/N was varied between the initial value, at which no annoyance was audible, down to a final value, where the audio signal became unusable(i.e. subjectively degraded and distorted or the speech became unintelligible).

In Gaussian Channel we needed 11 steps in a C/N range of 5.5 dB to go from a perfect quality down to a complete service breakdown (see the tables below). Similarly, for the Rural/Urban model we used 12 steps in a range of 11 dB. The absolute C/N values depend on the chosen Protection Level (PL).

Gaussian Channel configurations

	PL = 1 = 0.35	PL = 3 = 0.5	PL = 5 = 0.75
Step 0	C/N = 6.9 dB	C/N = 8.9 dB	C/N = 12.9 dB
Step 1	C/N = 6.4 dB	C/N = 8.4 dB	C/N = 12.4 dB
Step 2	C/N = 5.9 dB	C/N = 7.9 dB	C/N = 11.9 dB
Step 3	C/N = 5.4 dB	C/N = 7.4 dB	C/N = 11.4 dB
Step 4	C/N = 4.9 dB	C/N = 6.9 dB	C/N = 10.9 dB
Step 5	C/N = 4.4 dB	C/N = 6.4 dB	C/N = 10.4 dB
Step 6	C/N = 3.9 dB	C/N = 5.9 dB	C/N = 9.9 dB
Step 7	C/N = 3.4 dB	C/N = 5.4 dB	C/N = 9.4 dB
Step 8	C/N = 2.9 dB	C/N = 4.9 dB	C/N = 8.9 dB
Step 9	C/N = 2.4 dB	C/N = 4.4 dB	C/N = 8.4 dB
Step 10			C/N = 7.9 dB
Step 11			C/N = 7.4 dB

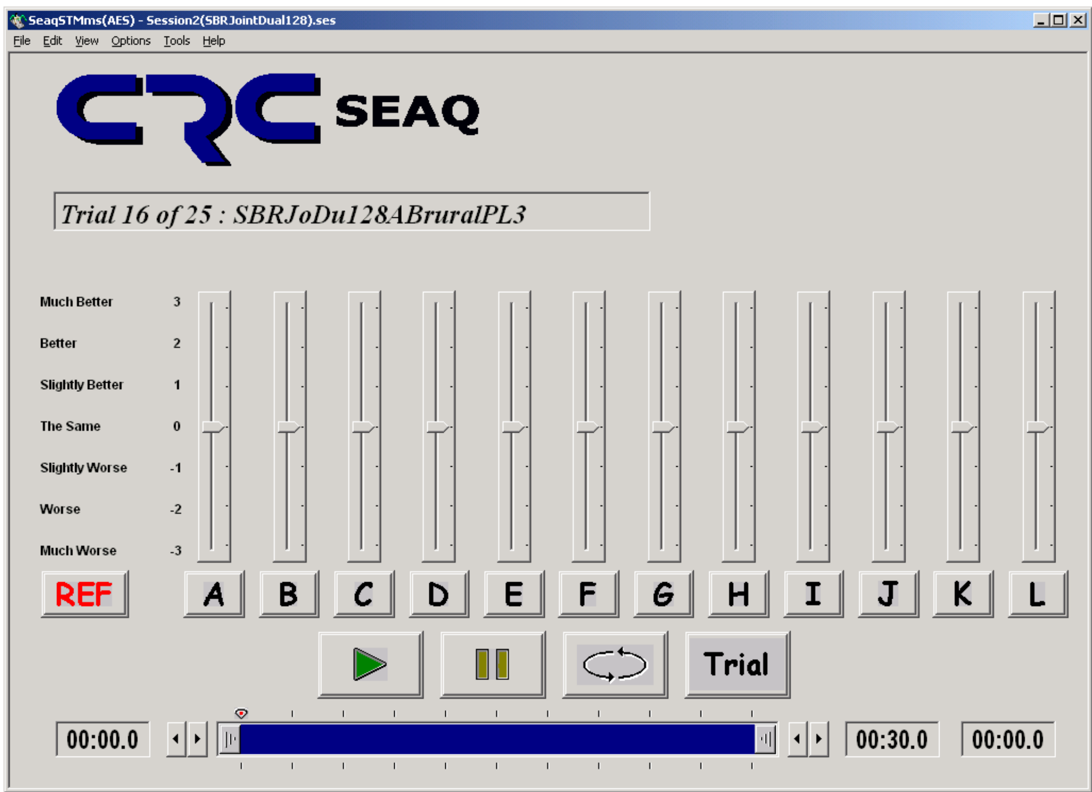
Rayleigh (Rural/Urban) Channel configurations

	SBR-Layer2plus PL = 3 = 0.5	Normal-Layer2 PL = 3 = 0.5
Step 1	C/N = 20.9 dB	
Step 2	C/N = 19.9 dB	
Step 3	C/N = 18.9 dB	C/N = 18.9 dB
Step 4	C/N = 17.9 dB	C/N = 17.9 dB
Step 5	C/N = 16.9 dB	C/N = 16.9 dB
Step 6	C/N = 15.9 dB	C/N = 15.9 dB
Step 7	C/N = 14.9 dB	C/N = 14.9 dB
Step 8	C/N = 13.9 dB	C/N = 13,9 dB
Step 9	C/N = 12.9 dB	C/N = 12.9 dB
Step 10	C/N = 11.9 dB	C/N = 11.9 dB
Step 11	C/N = 10.9 dB	C/N = 10.9 dB
Step 12	C/N = 9.9 dB	C/N = 9.9 dB
Step 13		C/N = 8.9 dB
Step 14		C/N = 7.9 dB

In total, we produced about 900 „mp2“ audio sequences, each of one minute duration. These were subsequently converted into linear „wav“ files using a Coding Technologies decoder software.

3 Description of tests

For our subjective tests a CRC SEAQ interface was used (see the following figure). The scale used was a 7-step scale (according to ITU-R BS 562-3) with a balanced range from +3 to –3 around the 0 value.



User interface SW CRC-SEAQ subjective test 7-point scale according to ITU-R BS 562-3.

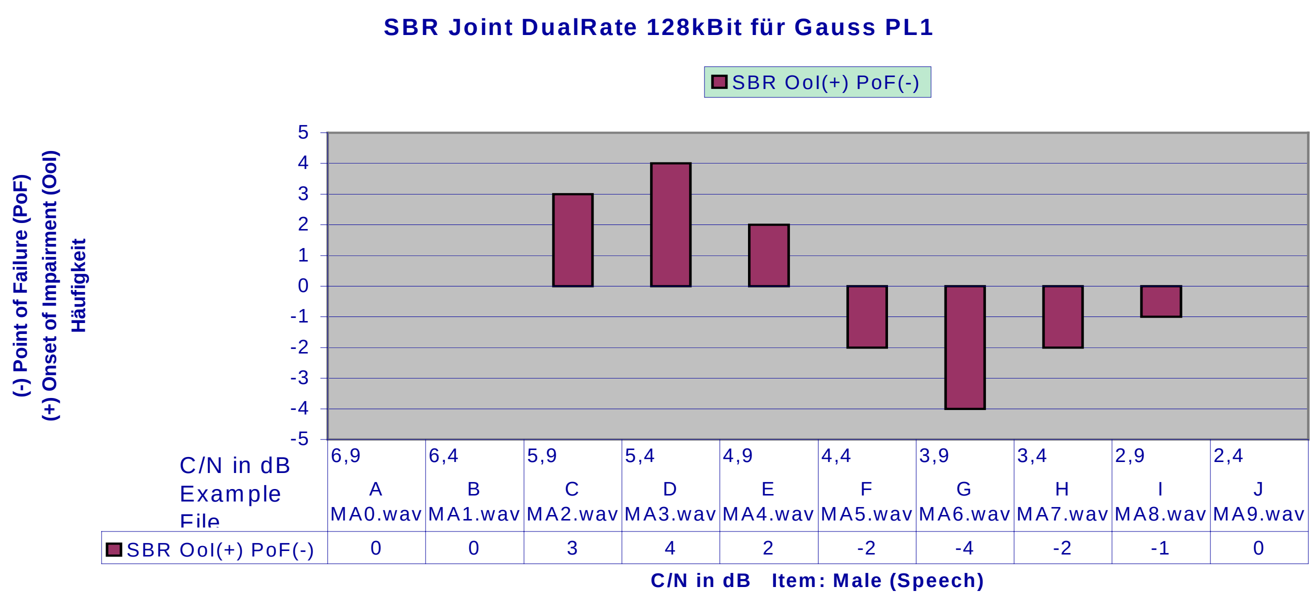
This interface contains 12 faders (A,B,...K,L), representing successive decreasing C/N and therefore decreasing audio quality. "A“ represents a signal with no annoyance and "L“ a totally disturbed audio signal.

The Coding Technologies decoding software uses an error concealment technique, whereas the Philips decoder which was used in the earlier BBC tests uses signal muting. To this end, test conditions had to be changed accordingly. The test persons did not have to count the number of signal mutings within an agreed space of time anymore but only had to answer simply the following questions:

- at which C/N ratio a slight annoyance of the audio signal first became perceptible. This point is called **Onset of Impairment (OOI)**
- at which C/N ratio the audio signal became totally disturbed, so that further listening became undesirable or impossible. This point is called **Point of Failure (PoF).**

The test persons were requested to shift up the fader into position +3 for Onset of Impairment (OOI) or alternatively shift down the fader into position –3 at Point of Failure (PoF).

As an example the result of 9 test persons ("Versuchspersonen") representing the occurrence of answers referring to Onset of Impairment (OoI) and Point of Failure (PoF) is shown in the following diagram:



By using the MUSHRA method, it was possible to carry out *individual* subjective tests. By contrast, the BBC tests were conducted as a *group* voting, which was prone to the "group" effect. The stability and accuracy of our tests is no doubt higher than that of the BBC, because the test subjects could vote independently (no interpersonal interaction and mutual influence was possible).

There were 8 sessions in which we tested at different bitrates, channel-modes and protection levels. Two sessions were carried out with Normal Layer 2 coded audio signals, the remaining six sessions were carried out with SBR Layer2plus.

Because of the large number of tests which required about 2 hours to complete, we divided the subjective test sessions into several blocks, so that a single test was generally not longer than half an hour.

	Session 0	Session 1	Session 2	Session 3	Session 4	Session 5	Session 6	Session 7
DataRate	Normal -L2 192 kBit	SBR-L2 160 kBit	SBR-L2 128 kBit	SBR-L2 96 kBit	SBR-L2 80 kBit	SBR-L2 64 kBit	SBR-L2 48 kBit	Normal -L2 48 kBit
Number	25	5	25	5	5	5	9	9
Examples	AB,CL,KL, FR,MA	AB,CL,KL, FR,MA	AB,CL,KL, FR,MA	AB,CL,KL, FR,MA	AB,CL,KL FR,MA	AB,CL,KL FR,MA	AB,CL,KL FR,MA (Rural/Urban only FR,MA)	AB,CL,KL FR,MA (Rural/Urban only FR,MA)
Channel-Mode	Gauss Rural/Urban	Gauss	Gauss Rural/Urban	Gauss	Gauss	Gauss	Gauss Rural/Urban	Gauss Rural/Urban
Protection-Level	PL1/PL3/PL5 (Rural/Urban only PL3)	PL3	PL1/PL3/PL5 (Rural/Urban only PL3)	PL3	PL3	PL3	PL3	PL3
Time about	36 min	6 min	36 min	6 min	6 min	6 min	11 min	11 min

Subjective hearing test Layer 2-SBR-DAB, 9 test persons at SBR-Layer2, 8 test persons at Normal-Layer2

4 Discussions of Results

The following table describes more distinctly the chosen configurations of the 8 test sessions (Session 0...7). As test signal 3 music examples (AB=Abba, CL=clarinet, KL=Glockenspiel) and 2 (“Sprache”) speech examples (FR=Frau=female voice, MA=Mann=male voice) were used.

DataRate	Function	Channel-Mode	Protection Level Code Rate	Test	Audio Example
DAB-SBR-Test	SBR-Layer2plus				
160 kbit/s	Stereo Single Rate	Gauss	PL3 = 0,5	Session 1	AB/CL/FR/KL/MA
128 kbit/s	Joint Dual Rate	Gauss Gauss Gauss Rural Urban	PL1 = 0,35 PL3 = 0,5 PL5 = 0,75 PL3 = 0,5 PL3 = 0,5	Session 2	AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA
96 kbit/s	Joint Dual Rate Low Comp	Gauss	PL3 = 0,5	Session 3	AB/CL/FR/KL/MA
80 kbit/s	Mono Single Rate	Gauss	PL3 = 0,5	Session 4	AB/CL/FR/KL/MA
64 kbit/s	Mono DualRate LowComp	Gauss	PL3 = 0,5	Session 5	AB/CL/FR/KL/MA
48 kbit/s	Mono DualRate LowComp	Gauss Rural Urban	PL3 = 0,5 PL3 = 0,5 PL3 = 0,5	Session 6	AB/CL/FR/KL/MA FR/MA FR/MA

As reference signal to the SBR-Layer2plus-coded audio signals the both following Normal-Layer2-coded signals were in use (192 kbit/s Joint Stereo sampling rate 48 kHz und 48 kbit/s Mono sampling rate 48 kHz).

DataRate	Function	Channel-Mode	Protection Level Code Rate	Test	Audio Example
DAB Reference	Normal-Layer2				
192kbit/s	Joint	Gauss Gauss Gauss Rural Urban	PL1 = 0,35 PL3 = 0,5 PL5 = 0,75 PL3 = 0,5 PL3 = 0,5	Session 0	AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA AB/CL/FR/KL/MA
48 kbit/s	Mono	Gauss Rural Urban	PL3 = 0,5 PL3 = 0,5 PL3 = 0,5	Session 7	AB/CL/FR/KL/MA FR/MA FR/MA

The complete individual results of all sessions are shown in the diagrams of the **Appendix 1**.

As an example the following configuration is depicted here:

Comparison channel-simulations (Gauss/Rural/Urban), measured by IRT, between SBR-Layer2plus and Normal-Layer2 at Protection Level PL=3 coded with high (128/192 kbit/s) and low (48 kbit/s) bitrate.

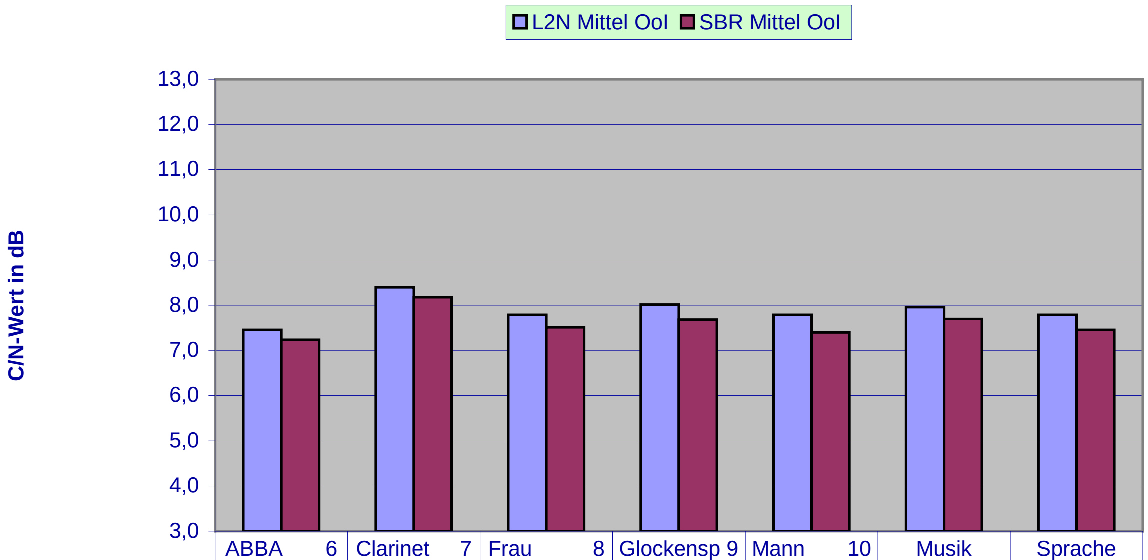
In the following diagrams are shown the result of each item (ABBA, Clarinet, Glockenspiel, Frau=female voice, Mann=male voice) and the average values (“Mittelwerte”) of the 3 music-items (“Musik”) and the 2 speech-items (“Sprache”)

In order to come to the point, music examples are showing with about 2.5 dB a much greater margin of deviation than speech examples. Clarinet belongs next to the “Glockenspiel” to the most critical signal of this subjective test, which should from the experiences of perceptual audio coding not be an amazing fact. This tendency is continuously proceeded for all test configurations (Data Rate, Protection Level, Channel Modes).Considered at an average value the difference between music- and speech-examples is in Gaussian-channel less than 0.5 dB, in Rayleigh-channel less than 1.5 dB.

The C/N ratio of all audio examples is under „Rural“-condition of the Rayleigh channel about 1.5 dB more critical than under „Urban“ condition.

Under Gaussian conditions the overall view seems to show a slight improvement in C/N-ratio (0 – 0,5 dB) at all data rates by using the SBR-Layer2plus-coding, under Rayleigh-channel conditions our subjective test lead to different results (high bitrate: improvement up to 1.8 dB in C/N-ratio, low bitrate a little loss of maximal 0.8 dB).

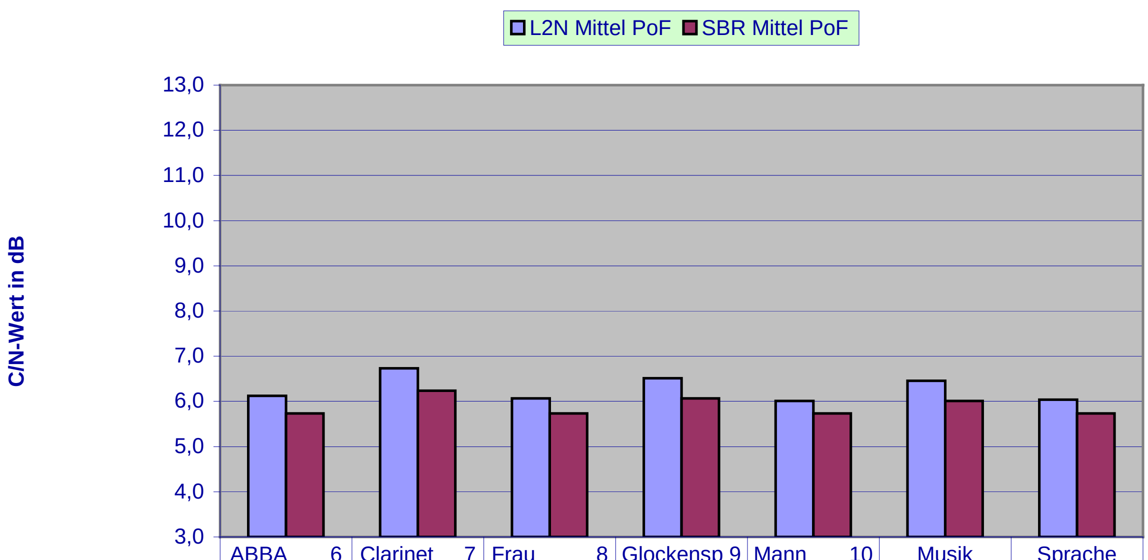
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL3



	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
L2N Mittel OoI	7,5	8,4	7,8	8,0	7,8	8,0	7,8
SBR Mittel OoI	7,2	8,2	7,5	7,7	7,4	7,7	7,5

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

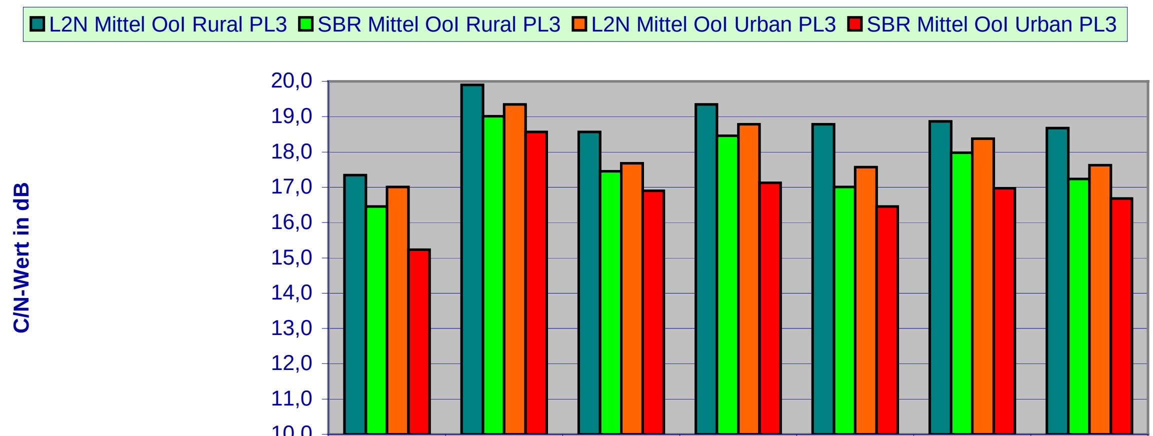
Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL3



	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
L2N Mittel PoF	6,1	6,7	6,1	6,5	6,0	6,5	6,0
SBR Mittel PoF	5,7	6,2	5,7	6,1	5,7	6,0	5,7

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

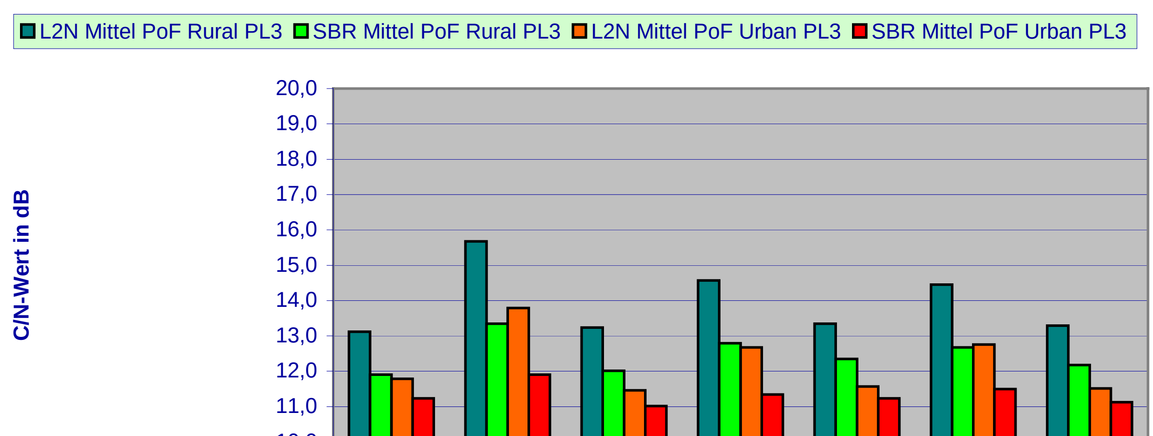
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Rural und Urban PL3



	ABBA 21	Clarinet 22	Frau 23	Glockensp 24	Mann 25	Musik	Sprache
L2N Mittel OoI Rural PL3	17,3	19,9	18,6	19,3	18,8	18,9	18,7
SBR Mittel OoI Rural PL3	16,5	19,0	17,5	18,5	17,0	18,0	17,2
L2N Mittel OoI Urban PL3	17,0	19,3	17,7	18,8	17,6	18,4	17,6
SBR Mittel OoI Urban PL3	15,2	18,6	16,9	17,1	16,5	17,0	16,7

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Rural und Urban PL3

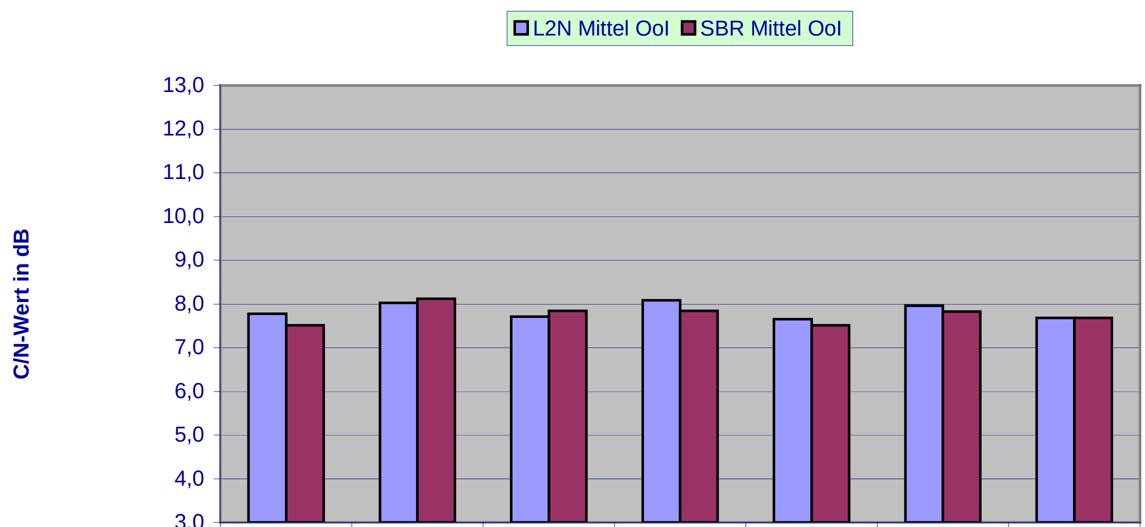


	ABBA 21	Clarinet 22	Frau 23	Glockensp 24	Mann 25	Musik	Sprache
L2N Mittel PoF Rural PL3	13,1	15,7	13,2	14,6	13,3	14,5	13,3
SBR Mittel PoF Rural PL3	11,9	13,3	12,0	12,8	12,3	12,7	12,2
L2N Mittel PoF Urban PL3	11,8	13,8	11,5	12,7	11,6	12,8	11,5
SBR Mittel PoF Urban PL3	11,2	11,9	11,0	11,3	11,2	11,5	11,1

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Low bitrate-coding (at 48 kbit/s) is leading to the following results:

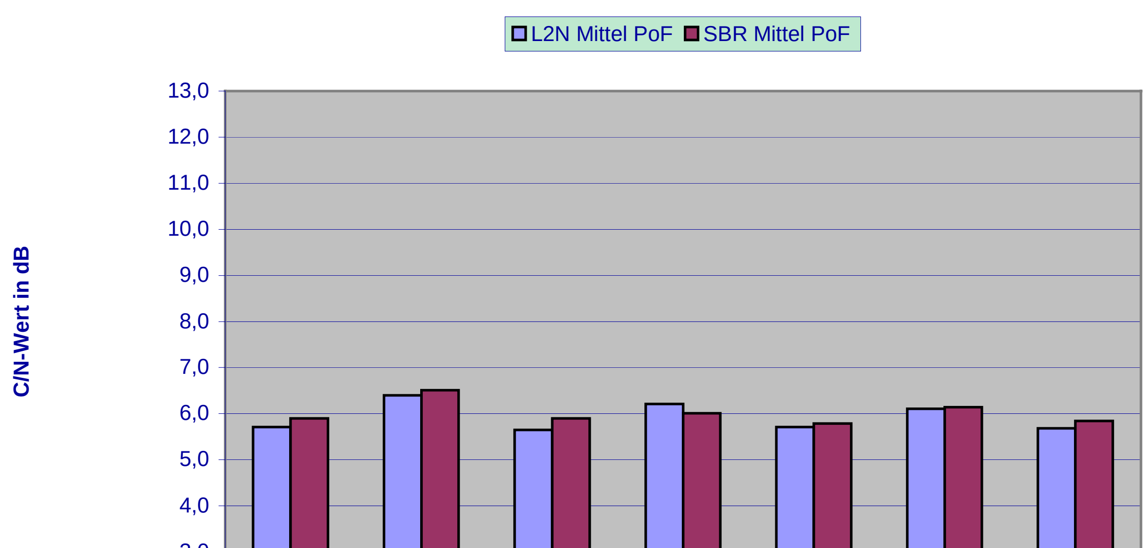
Onset of Impairment C/N-Mittelwerte L2Normal und SBR Mono48kBit Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel OoI	7,8	8,0	7,7	8,1	7,7	8,0	7,7
SBR Mittel OoI	7,5	8,1	7,8	7,8	7,5	7,8	7,7

Onset of Impairment 5 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

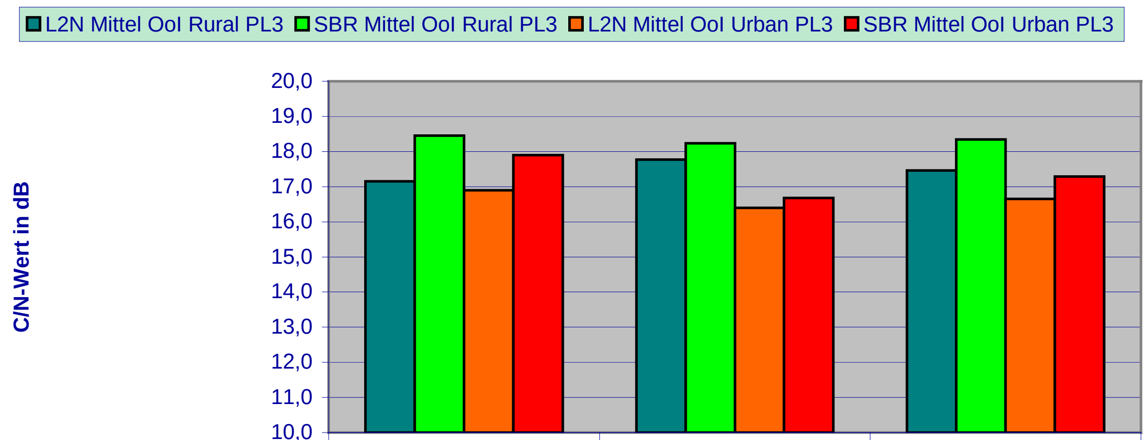
Point of Failure C/N-Mittelwerte L2Normal und SBR Mono48kBit Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel PoF	5,7	6,4	5,7	6,2	5,7	6,1	5,7
SBR Mittel PoF	5,9	6,5	5,9	6,0	5,8	6,1	5,8

Point of Failure 5 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

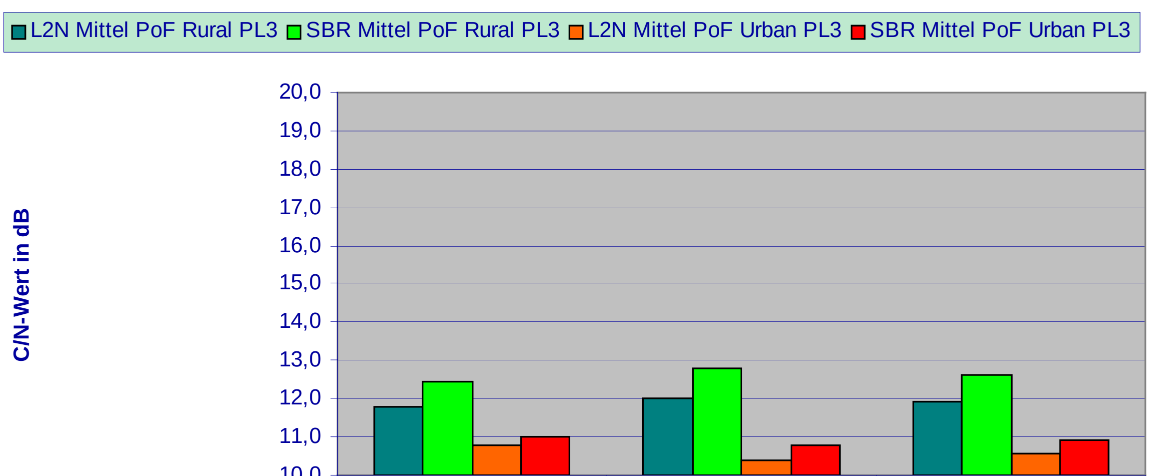
Onset of Impairment C/N-Mittelwerte L2Normal und SBR
Mono48kBit Rural und Urban PL3



	Frau 8	Mann 9	Sprache
L2N Mittel OoI Rural PL3	17,2	17,8	17,5
SBR Mittel OoI Rural PL3	18,5	18,2	18,3
L2N Mittel OoI Urban PL3	16,9	16,4	16,7
SBR Mittel OoI Urban PL3	17,9	16,7	17,3

Onset of Impairment 2 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

Point of Failure C/N-Mittelwerte L2Normal und SBR
Mono48kBit Rural und Urban PL3



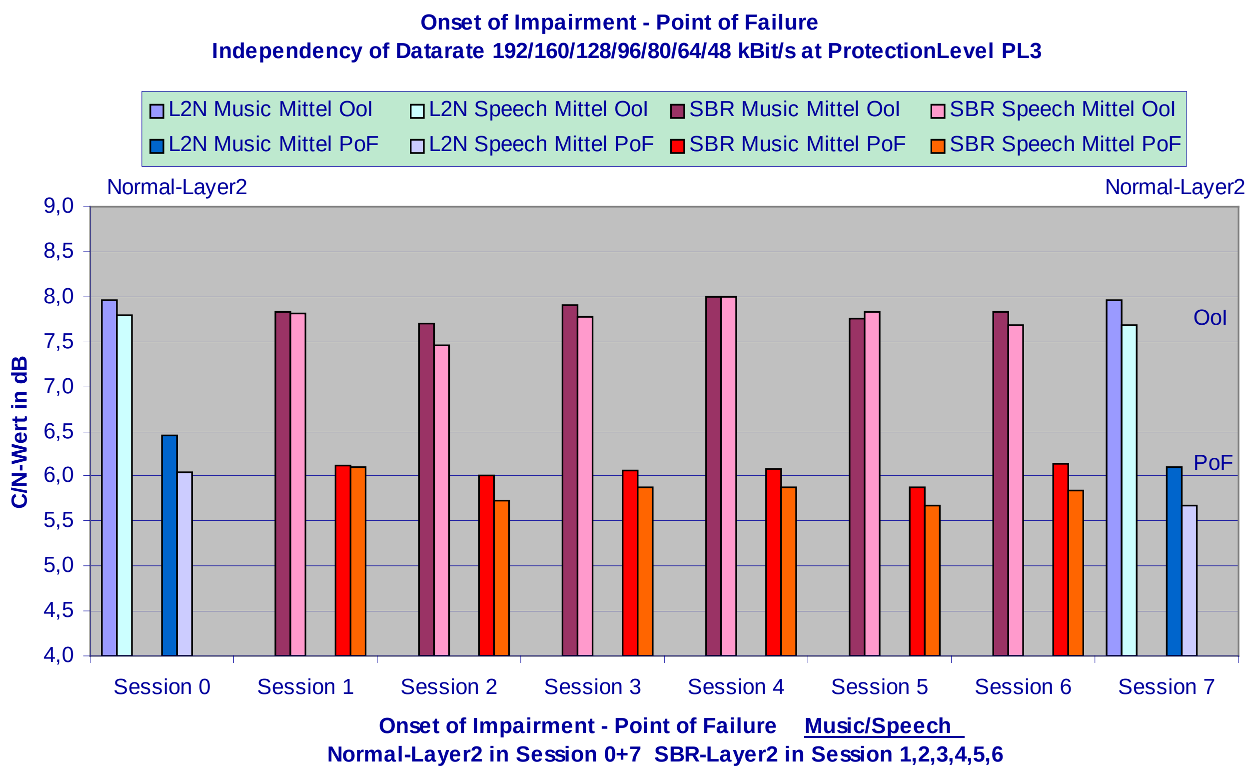
	Frau 8	Mann 9	Sprache
L2N Mittel PoF Rural PL3	11,8	12,0	11,9
SBR Mittel PoF Rural PL3	12,5	12,8	12,6
L2N Mittel PoF Urban PL3	10,8	10,4	10,6
SBR Mittel PoF Urban PL3	11,0	10,8	10,9

Point of Failure 2 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

5 Comparisons of the IRT-Results with the earlier BBC-Results

The BBC studies from 1996 show that the OoI and PoF points do not depend on bit rate. We attempted to confirm (or not) this finding by testing the different bitrates between 192 and 48 kbit/s in Gaussian channel). We took two extreme conditions:

- high (192 kbit/s Normal-Layer2 respectively 128 kbit/s SBR-Layer2plus), and
- low (48 kbit/s Normal- und SBR Layer2plus) coding bitrate.



Channel-simulation of a DAB transmission was carried out in VHF Band III with audio examples of high and low bitrate at different Protection Level PL = 1 / 3 / 5:

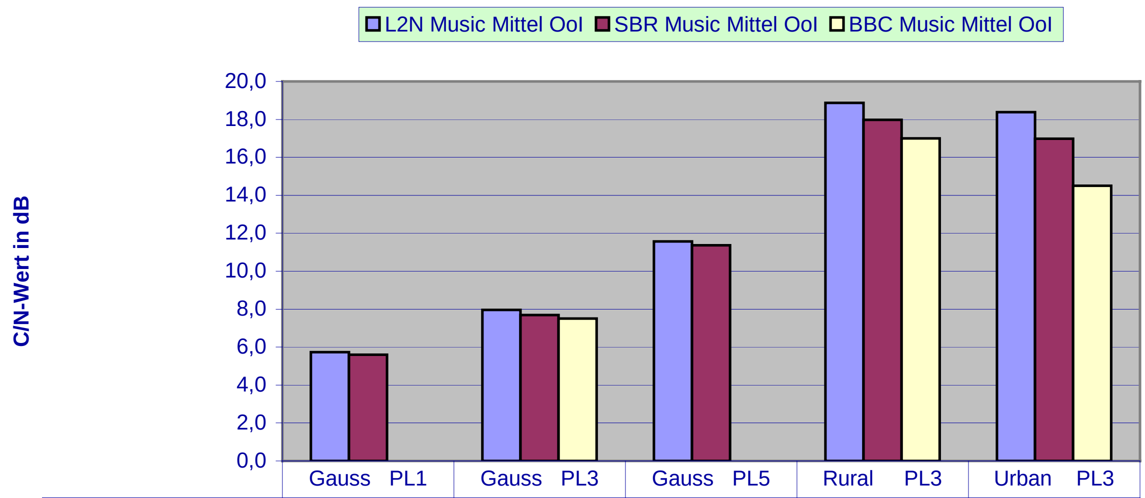
Onset of Impairment and **Point of Failure** at **high** bitrate for **Music** and **Speech** audio examples are compared in the following figures:

IRT: Normal-Layer2 Joint 192 kbit/s (L2N blue columns);

SBR-Layer2plus Joint Dual Rate 128 kbit/s (SBR red columns);

BBC: Normal-Layer2 Joint 224 kbit/s (BBC yellow columns,
but not all configurations tested at BBC);

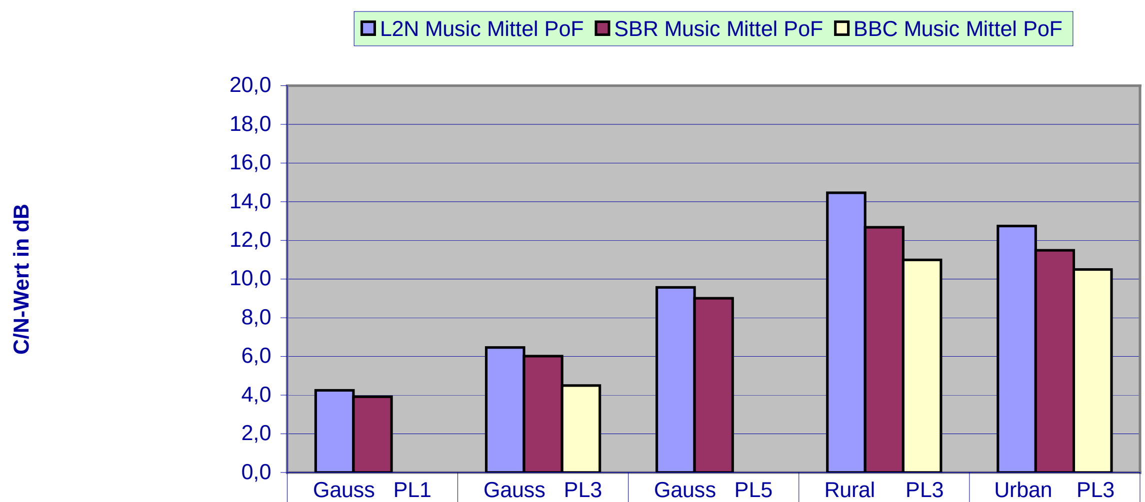
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit



	Gauss PL1	Gauss PL3	Gauss PL5	Rural PL3	Urban PL3
L2N Music Mittel OoI	5,7	8,0	11,6	18,9	18,4
SBR Music Mittel OoI	5,6	7,7	11,4	18,0	17,0
BBC Music Mittel OoI	0,0	7,5	0,0	17,0	14,5

Onset of Impairment <u>Music</u> 9 Versuchspersonen

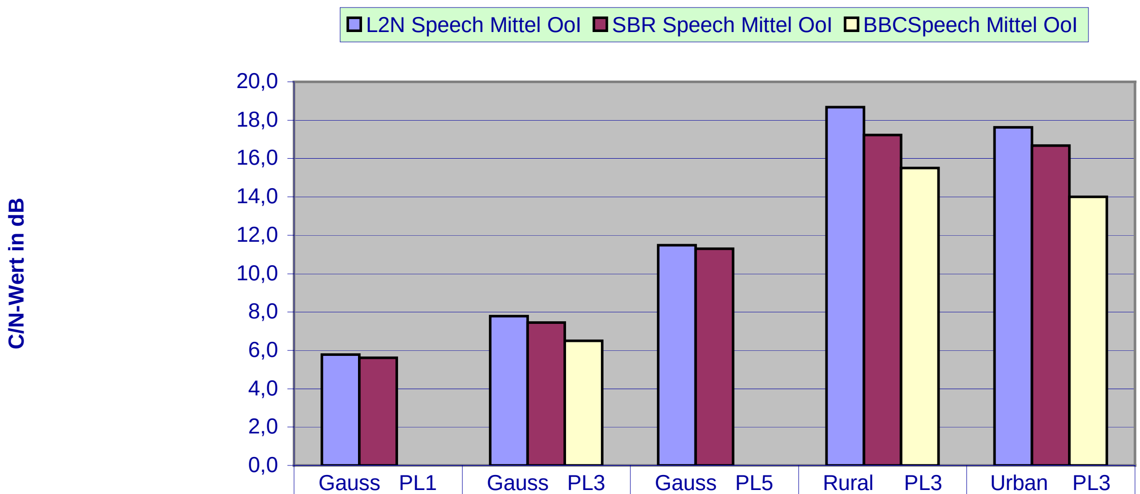
Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit



	Gauss PL1	Gauss PL3	Gauss PL5	Rural PL3	Urban PL3
L2N Music Mittel PoF	4,3	6,5	9,6	14,5	12,8
SBR Music Mittel PoF	3,9	6,0	9,0	12,7	11,5
BBC Music Mittel PoF	0,0	4,5	0,0	11,0	10,5

Point of Failure <u>Music</u> 9 Versuchspersonen

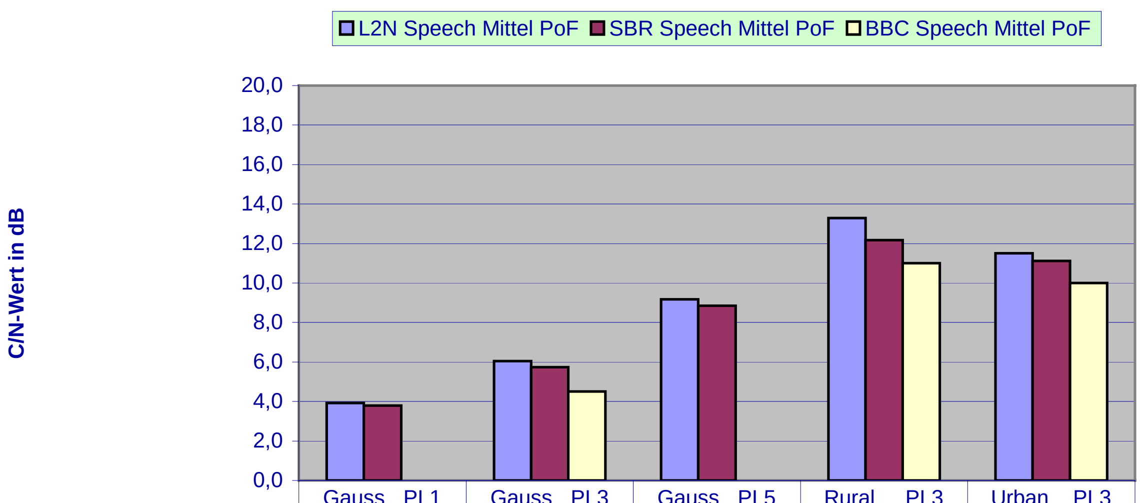
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit



	Gauss PL1	Gauss PL3	Gauss PL5	Rural PL3	Urban PL3
L2N Speech Mittel OoI	5,8	7,8	11,5	18,7	17,6
SBR Speech Mittel OoI	5,6	7,5	11,3	17,2	16,7
BBCSpeech Mittel OoI	0,0	6,5	0,0	15,5	14,0

Onset of Impairment Speech 9 Versuchspersonen

Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit



	Gauss PL1	Gauss PL3	Gauss PL5	Rural PL3	Urban PL3
L2N Speech Mittel PoF	3,9	6,0	9,2	13,3	11,5
SBR Speech Mittel PoF	3,8	5,7	8,8	12,2	11,1
BBC Speech Mittel PoF	0,0	4,5	0,0	11,0	10,0

Point of Failure Speech 9 Versuchspersonen

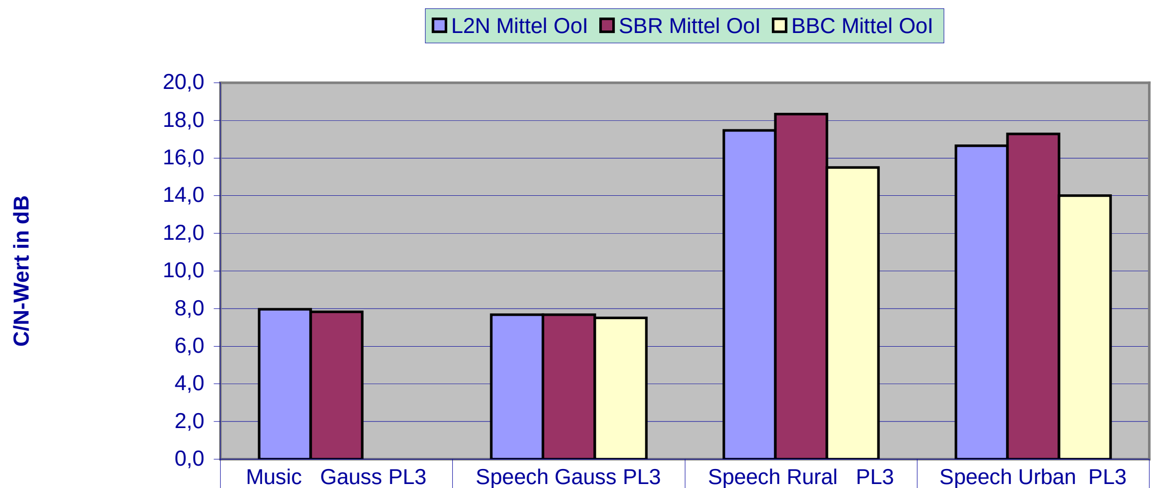
Channel-simulation of a DAB transmission with **low** bitrate was only carried out at Protection Level PL = 3:

IRT: Layer2Normal Mono 48 kbit/s;

SBR-L2+ Mono DualRate lowCompatibility 48 kbit/s;

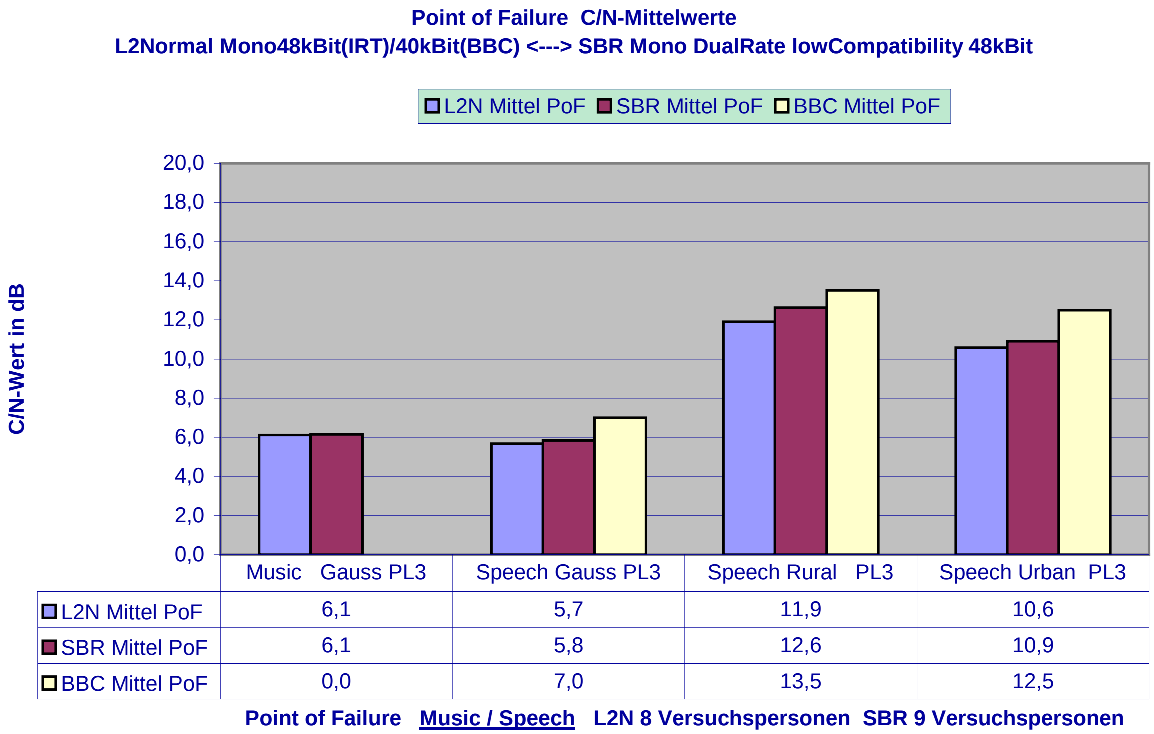
BBC: Layer2Normal Mono 40 kbit/s (BBC did not tested Music audio examples);

Onset of Impairment C/N-Mittelwerte
L2Normal Mono48kBit(IRT)/40kBit(BBC) <---> SBR Mono DualRate lowCompatibility 48kBit



	Music Gauss PL3	Speech Gauss PL3	Speech Rural PL3	Speech Urban PL3
L2N Mittel OoI	8,0	7,7	17,5	16,7
SBR Mittel OoI	7,8	7,7	18,3	17,3
BBC Mittel OoI	0,0	7,5	15,5	14,0

Onset of Impairment Music / Speech L2N 8 Versuchspersonen SBR 9 Versuchspersonen



	Music Gauss PL3	Speech Gauss PL3	Speech Rural PL3	Speech Urban PL3
L2N Mittel PoF	6,1	5,7	11,9	10,6
SBR Mittel PoF	6,1	5,8	12,6	10,9
BBC Mittel PoF	0,0	7,0	13,5	12,5

From the above diagrams it is possible to deduce that our results are similar to the BBC ones, namely that the OoI and PoF points do not significantly depend of the bit rate used.

It is evident that if the protection decreases, higher C/N-ratio is necessary for an error-free performance.

Comparing the Gaussian channel as a reference, the mobile reception in a driving vehicle requires an increased C/N ratio. This finding complies with the BBC results which state that there is a slight degradation for "rural“ conditions compared to “urban“ conditions, given the same C/N.

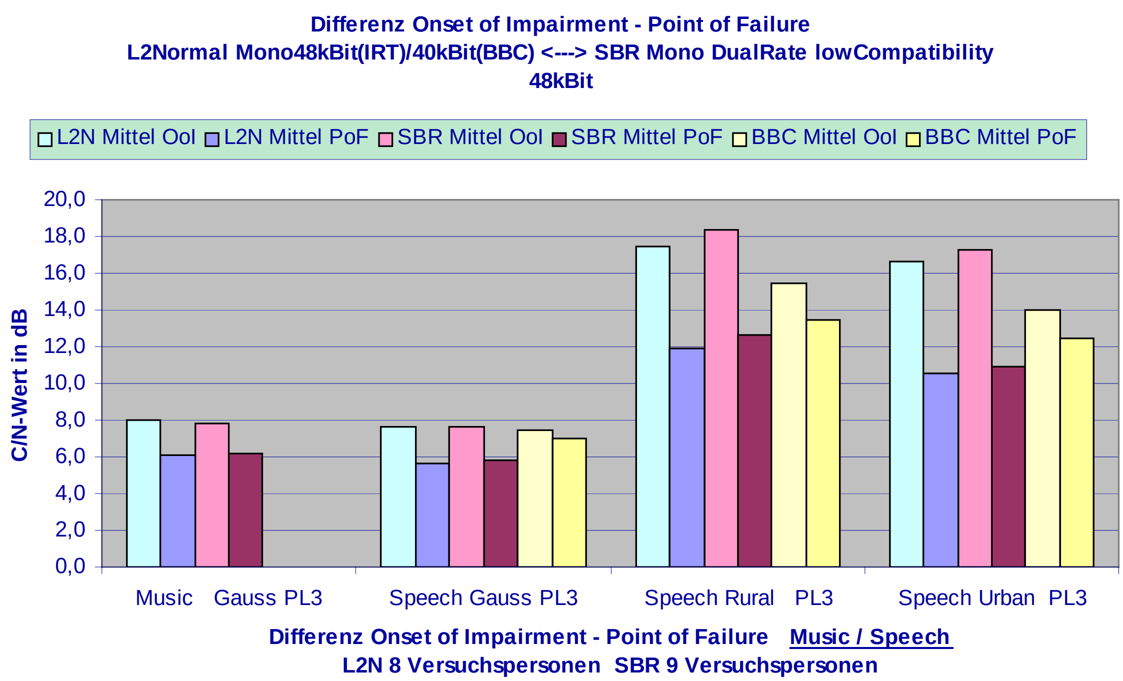
Our study showed that the difference between music and speech examples is less than 0.5 dB (for Gaussian channel). This compares favourably to the BBC finding, that in Rayleigh channel this difference is less that 1.5 dB.

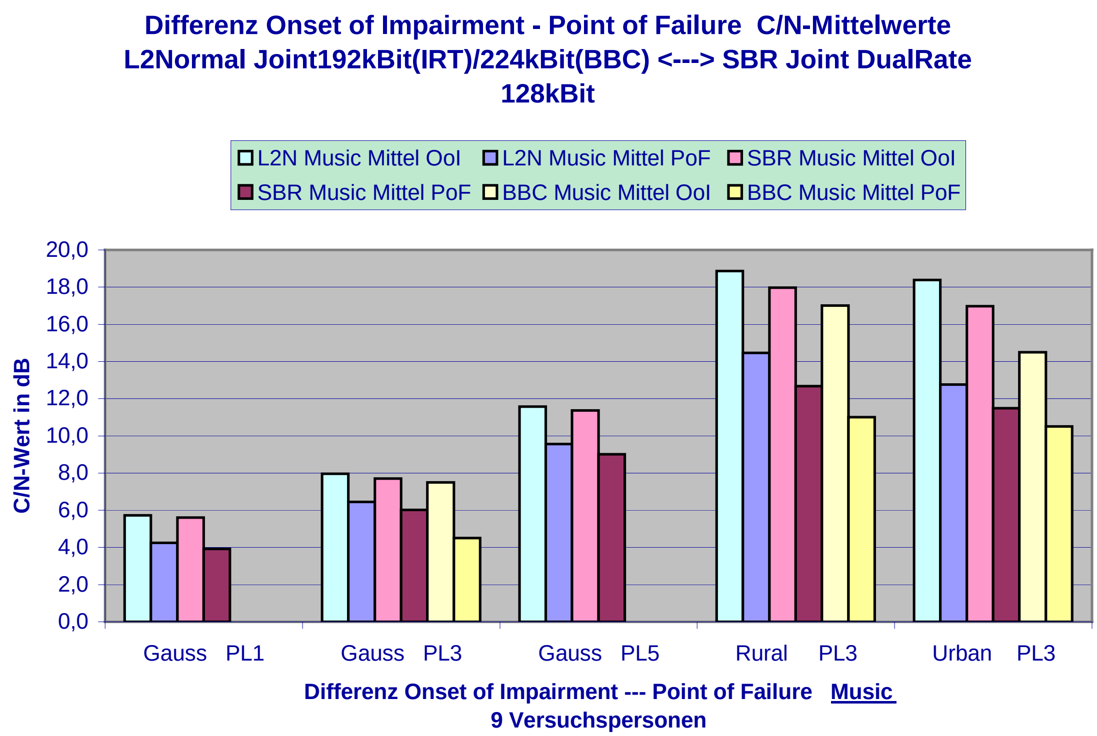
For high bitrates, SBR at 128 kbit/s gives an improvement of up to 1.8 dB in C/N-ratio compared to Normal-Layer2 at 192 kbit/s. At low bitrates (e.g. 48 kbit/s) the reverse seems to be the case: SBR requires 0.8 dB more than Layer 2.

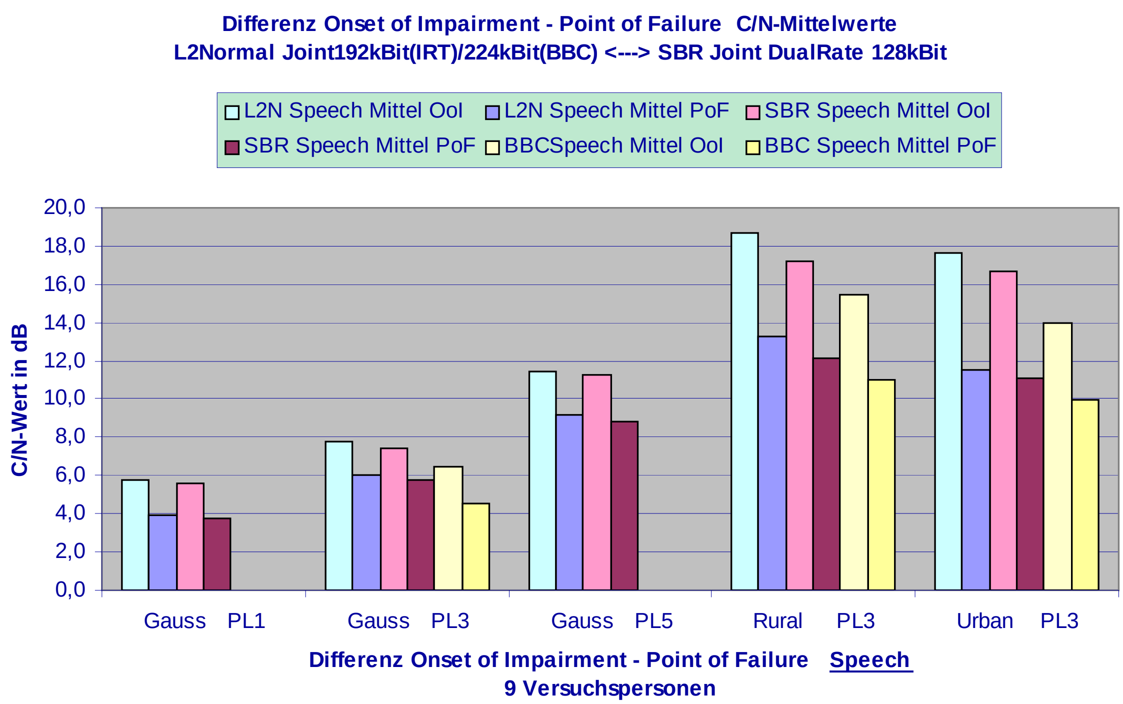
In general, it is very difficult to compare the BBC results with ours, because different decoders were use then and now. The BBC used Philips DAB3 [3] or DAB452-Decoders [4]. IRT however conducted their tests using the Coding Technologies encoder/decoder software. Differences may arise also from the different error correction techniques used now and then (concealment versus muting).

6 Conclusions on Part 1 (Error Sensitivity)

The following diagrams compare at the both in the test preferred bitrates (low: IRT-SBR and IRT-L2N = 48 / BBC-L2N = 40 kbit/s; high: IRT-SBR = 128 / IRT-L2N = 192 / BBC-L2N = 224 kbit/s) at an average value over music- and speech-examples the Onset-of-Impairment and the Point-of-Failure, so called "difference".




Differenz Onset of Impairment - Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit
L2N Music Mittel OoI
L2N Music Mittel PoF
SBR Music Mittel OoI
SBR Music Mittel PoF
BBC Music Mittel OoI
BBC Music Mittel PoF
C/N-Wert in dB
20,0
18,0
16,0
14,0
12,0
10,0
8,0
6,0
4,0
2,0
0,0
Gauss PL1
Gauss PL3
Gauss PL5
Rural PL3
Urban PL3
Differenz Onset of Impairment --- Point of Failure Music
9 Versuchspersonen


Differenz Onset of Impairment - Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit(IRT)/224kBit(BBC) <---> SBR Joint DualRate 128kBit
L2N Speech Mittel OoI
L2N Speech Mittel PoF
SBR Speech Mittel OoI
SBR Speech Mittel PoF
BBCSpeech Mittel OoI
BBC Speech Mittel PoF
C/N-Wert in dB
20,0
18,0
16,0
14,0
12,0
10,0
8,0
6,0
4,0
2,0
0,0
Gauss PL1
Gauss PL3
Gauss PL5
Rural PL3
Urban PL3
Differenz Onset of Impairment - Point of Failure Speech
9 Versuchspersonen

Considering the above graphs, the following general conclusions can be made:

The present IRT studies show that the previous BBC results about the error performance of the DAB transmission system give similar results. The difference between Onset of Impairment and Point of Failure amounts at Protection Level 1 and 3 to about 2 dB, and at Protection Level 5to about 3 dB. The difference between "rural“ and "urban“ (in terms of C/N required) is about 1.5 dB. The transmission errors perceived for music signals are about 0.5 dB more critical than for speech signals.

The impact of SBR coding can be summarized as follows:

- at high audio bitrates (192 kbit/s) SBR improves C/N by 1.0-1.5 dB for "rural" and 0.5-1.0 dB for "urban".
- at low bit rates (e.g. 48 kbit/s) SBR causes C/N degradation of 0.75 dB for "rural" and 0.4 dB for "urban".

The degradation at low bitrates can probably be attributed to the fact, that the better coding at SBR leads to a more precise detection of transmission errors. In order to compensate for these errors, a higher C/N is required. Similarly, one could argue that a relatively poor quality Layer 2 system masks errors and therefore requires lower C/N.

Our study shows that the coverage area would not change significantly if SBR is introduced in DAB. In other words, the RF performance in terms of C/N will not be significantly improved or degraded if SBR is introduced to DAB. The existing and new receivers could be used in the same area. From this point of view, the backward compatibility criterion could be fulfilled.

Part 2: Coding Gain and Compatibility

7 Scope of Part 2: Coding gain and Compatibility

The aim of Part 2 is to determine the amount of bitrate saving that MPEG Layer II^{SBR} codec gives over the standard MPEG Layer II codec without loss of audio quality. To determine this coding gain a diploma thesis [1] was written by IRT. Subjective listening tests were carried out with 6 test sequences at different bitrates and stereo modes (stereo, joint-stereo and mono). In all, 17 participants were used in the tests.

7.1 Audio Test Sequences

The following test items were chosen:

1. **Cembalo (harpsichord)** (9 sec.) sequence of single tones over 3 octaves, SQAM Test CD, Track 40
2. **Klassischer Konzertausschnitt (classical music sequence)** (14 sec.) Brahms Symphony Nr.1 c-minor
3. **Popmusik** (15 sec.) rhythmical sequence, keyboards, guitar, percussion, drumloop,
4. **Sprachsignal (speech)** (19 sec.) male speaker
5. **Sologesang (solo voice)** (10 sec.) voice female Suzanne Vega, CD Solitude Standig, Track 1
6. **Sportstadion (sports stadium)** (12 sec.) ice hockey match, IRT-production, demonstration material

7.2 Bitrates and Stereo Modes

A range of bitrates at different stereo modes for all the test items were chosen to give a useful comparison of the audio quality between the standard MPEG Layer II codec and the MPEG Layer II^{SBR} version. It soon became apparent that the coding gain achievable with SBR fell between two limits. The first limit is at high bitrates where the standard codec gives a sufficiently high audio quality. The second limit is at very low bitrates where the use of a standard codec in a DAB system transmitting SBR coded audio would lead to substantial loss in quality. However, such low bitrates are unlikely to be used in most public broadcast systems.

Bit rates used for coding gain investigation								
Bit rate	48 kbit/s	64 kbit/s	80 kbit/s	96 kbit/s	112 kbit/s	128 kbit/s	160 kbit/s	192 kbit/s
MPEG Layer II Joint Stereo			**X** (LSF)	**X** (LSF)		**X** (LSF)	**X**	**X**
Layer II^{SBR} Stereo		**X**	**X**	**X**	**X**	**X**	**X**	
Layer II^{SBR} Joint Stereo		**X**	**X**	**X**	**X**	**X**		
MPEG Layer II Mono	**X** (LSF)	**X** (LSF)	**X**	**X**				
Layer II^{SBR} Mono	**X**	**X**						

Used bit rates at SBR Layer II and Standard Layer II (LSF=half sampling frequency)

The range of bitrates chosen for MPEG Layer II^{SBR} was between 64 and 128kbit/s for stereo, and for mono 48 and 64kbit/s were chosen. The choice of bitrates for standard Layer II ranged from 80 to 192kbit/s in joint-stereo mode (the degradation in stereo imaging can be tolerated), which were chosen to give a similar quality range to the Layer II^{SBR} versions. For standard Layer II in mono the range of bitrates was 48 to 96kbit/s. For the lower bitrates (128kbit/s and below for joint-stereo) standard Layer II was used in “half sampling rate” mode (also known as LSF – low sampling frequency) which samples at 24kHz instead of 48kHz (according to the ITU MPEG 2 standard). The table above shows the chosen configurations.

7.3 Subjective Test Method

The subjective test method used was **MUSHRA** (**MU**lti **S**timulus test with **H**idden **R**eference and **A**nchor), which is well proven for the range of audio quality used in these tests.

The subjective scoring scale used ranges from 0 to 100, within which 5 classes of quality are described as “bad”, “poor”, “fair”, “good” and “excellent”.

7.4 Test Preparation

The audio test items were encoded in both standard Layer II and Layer II^{SBR} with software encoders developed by *Coding Technologies (MPEG-1/2 Layer II Demo Encoder V 1.5.0 Feb 26 2002* and *Layer 2 + SBR Demo Encoder V 0.7.0 July 25 2002)*. The software decoder used was also from *Coding Technologies* (*Layer II + SBR Demo Decoder V 0.6.0 July 15 2002).*

For the mono test items sampled at half rate (LSF), a software encoder developed by IRT was used (*MPEG-1/2 Layer 2 Demo Encoder V 1.5.0 February 26 2002)* which ran on a Silicon Graphics PC. The Layer II decoding was carried out with the *Soundapp v2.7.3* software running on an Apple Macintosh G4. The two low-pass filtered (7kHz and 3.5kHz) anchor items were generated using the *Cool Edit 2000* software.

7.5 MUSHRA software

The MUSHRA software was developed by Communications-Research-Centre Canada (*CRC-SEAQ Subjective Test Module, Version 1.18)* and ran on a Windows 2000 PC. In all, 17 people participated in the subjective listening tests.

8 Analysis and Results – *Quality* Gain

The analysis was based on the average values (with 95% confidence intervals) of the 100-point quality scale for each of the bitrates and coder configurations. The first stage of analysis was to determine the **quality gain,** then the next stage was to evaluate the **coding gain** from this.

The **quality gain** can be described as the perceived increase in quality of one codec over another at a given bitrate. The **coding gain** can be derived from the quality by identifying equivalent quality scores for each codec and working out the difference in bitrates required to achieve those scores.

8.1 Stereo Test Sequence Results

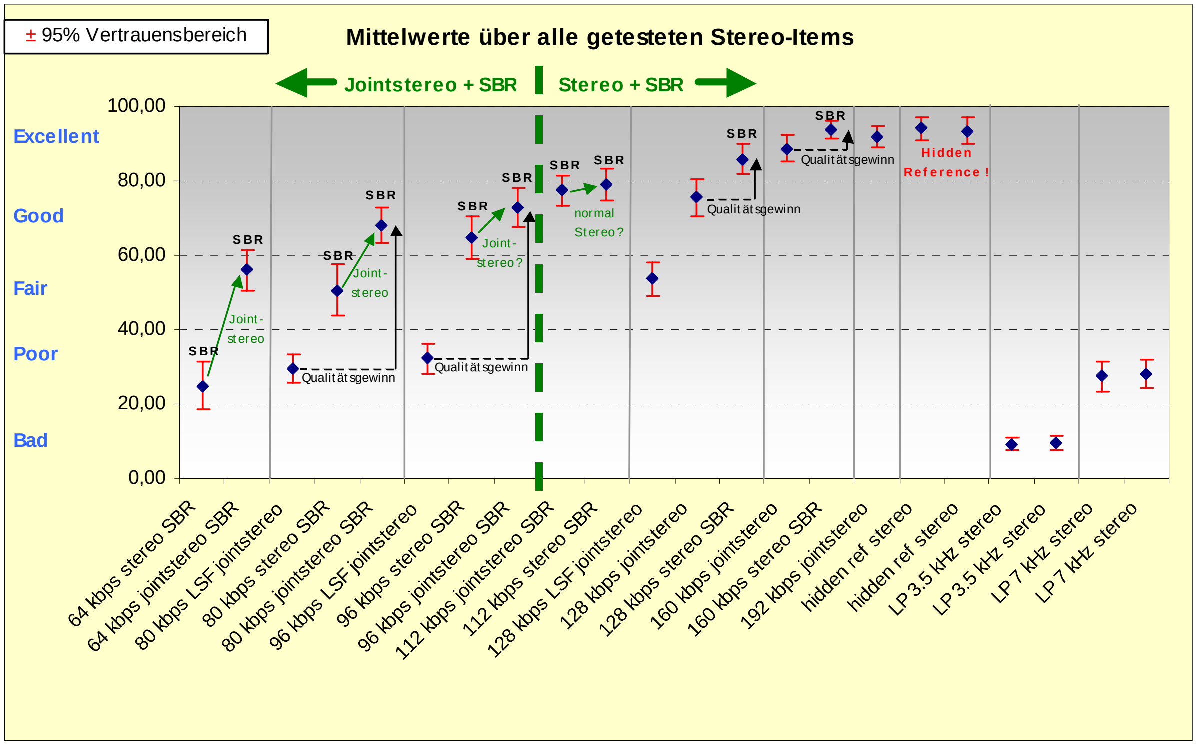
In the following sections the figures show the **quality gain** sorted in 3 different ways:

- a figure for each of the 6 test items
- a figure for each bitrate
- a figure for each stereo mode.

All the figures are showing similar performances for Layer II^{SBR} given these situations:

- at high bitrates the **quality gain** is minimal because standard Layer II provides sufficient quality
- at low bitrates the **quality gain** can be improved by using Layer II^{SBR} coding
- at high bitrates the difference in **quality gain** between joint-stereo and stereo coding decreases (applies to both standard and SBR coding).

8.1.1 Average over all stereo sequences



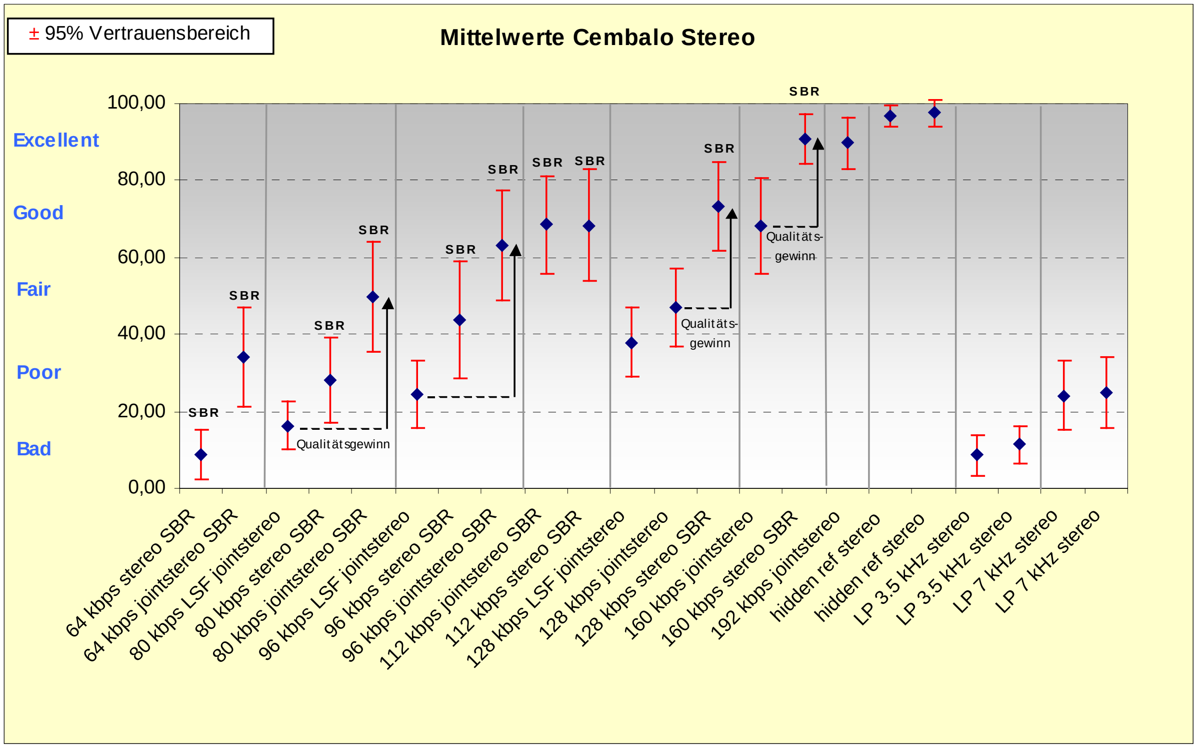
Average quality scores averaged over all stereo sequences

Figure above shows the average quality scores for each bitrate and codec type over all the test sequences for stereo. The **quality gain** SBR provides with bitrates in the range 80 to 160kbit/s is clearly apparent. At 80 kbit/s LSF joint-stereo standard Layer II is classed as "poor", whereas Layer II^{SBR} stereo is classed as "fair", and Layer II^{SBR} joint-stereo is "good". At 96kit/s a similar level of improvement is gained by Layer II^{SBR} over the standard Layer II.

In the range of bitrates between 64kbit/s and 112kbit/s Layer II^{SBR} joint-stereo gives better scores than Layer II^{SBR} stereo; however above 112kbit/s the difference between joint-stereo and stereo becomes less.

At 128kbit/s and above the quality gain between Layer II^{SBR} stereo and standard Layer II decreases because the standard Layer II provides sufficient quality. At 160 and 192kbit/s standard Layer II in stereo provided no statistically perceptible difference between the original and coded versions.

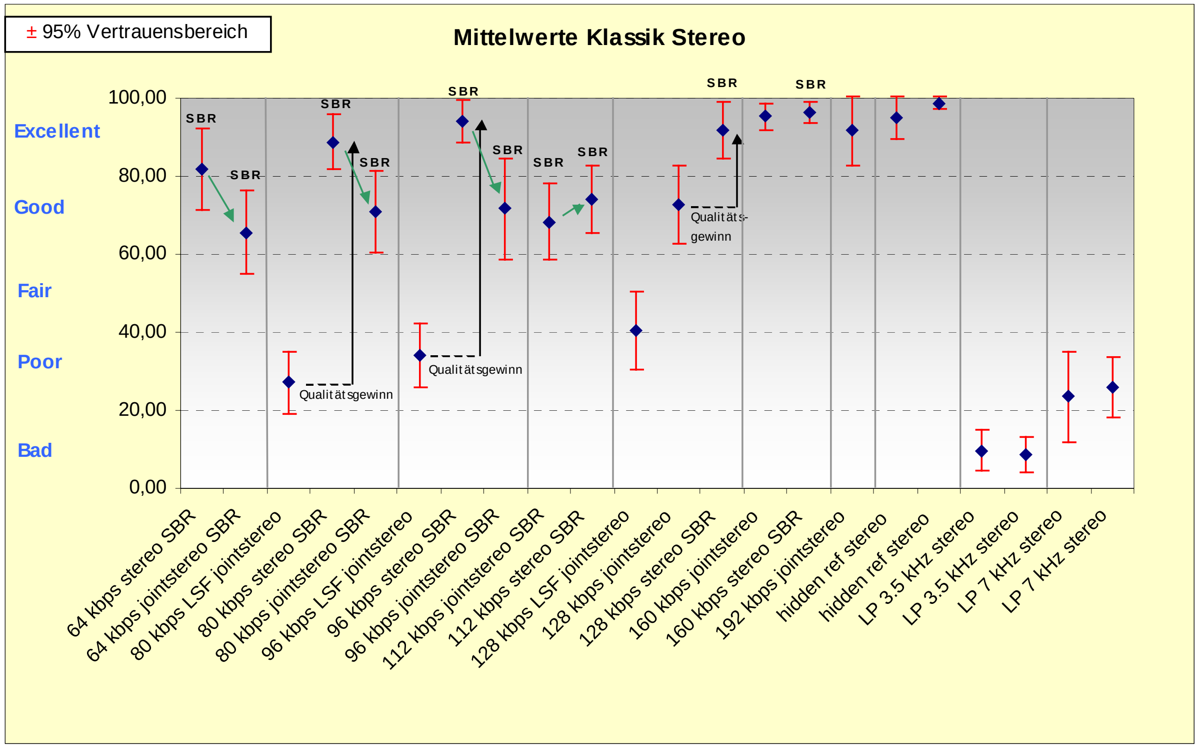
8.1.2 Stereo Harpsichord



Quality scores for harpsichord

The harpsichord is the most critical item, which shows the same relative quality as the other items over all the bitrates and coders, but at a lower absolute level. At 160kbit/s Layer II^{SBR} was distinguishable from the original signal, as was also a similar quality level to standard Layer II at 192kbit/s joint-stereo.

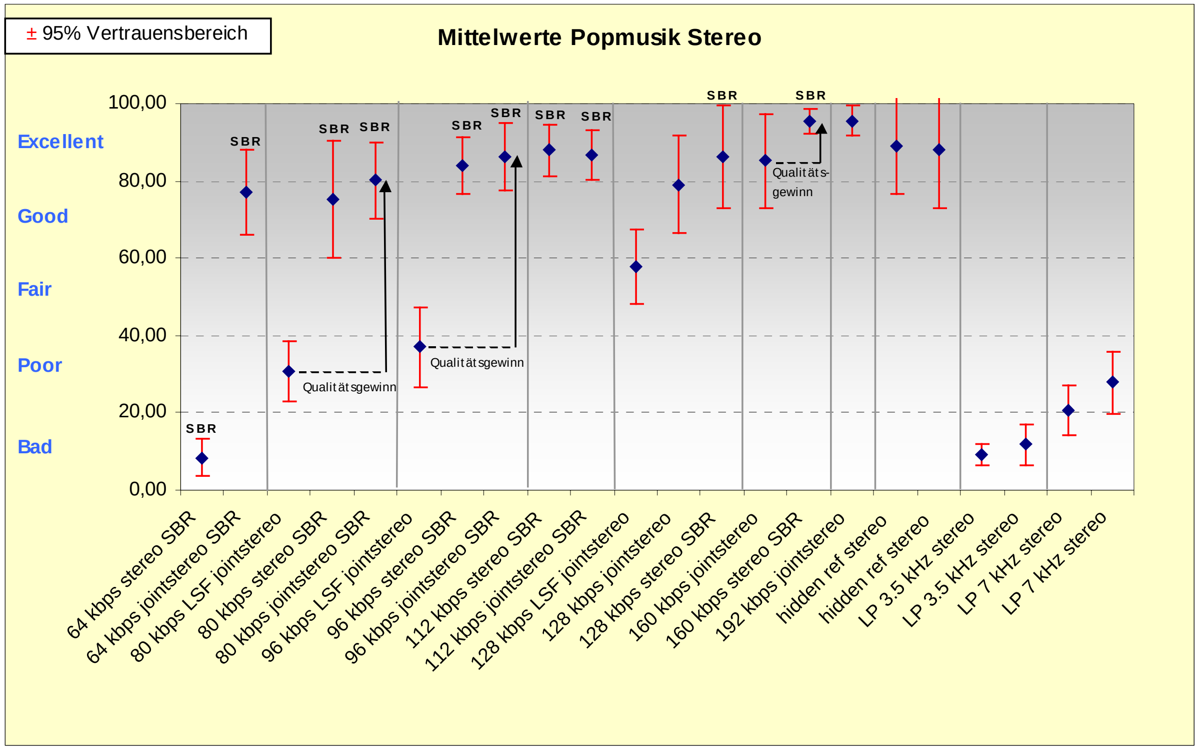
8.1.3 Stereo Classical Music



Quality scores for stereo classical music

The classical music item scores significantly differently to the other test items. The quality gain of Layer II^{SBR} over standard Layer II at 80, 96 and 128kbit/s is very large, jumping two classes (from “poor” to “excellent” in the 80kbit/s case). At 160kbit/s there was no improvement in quality by SBR. Classical music appears to be sensitive to shifts in the stereo imaging, which resulted in stereo giving better scores than joint-stereo.

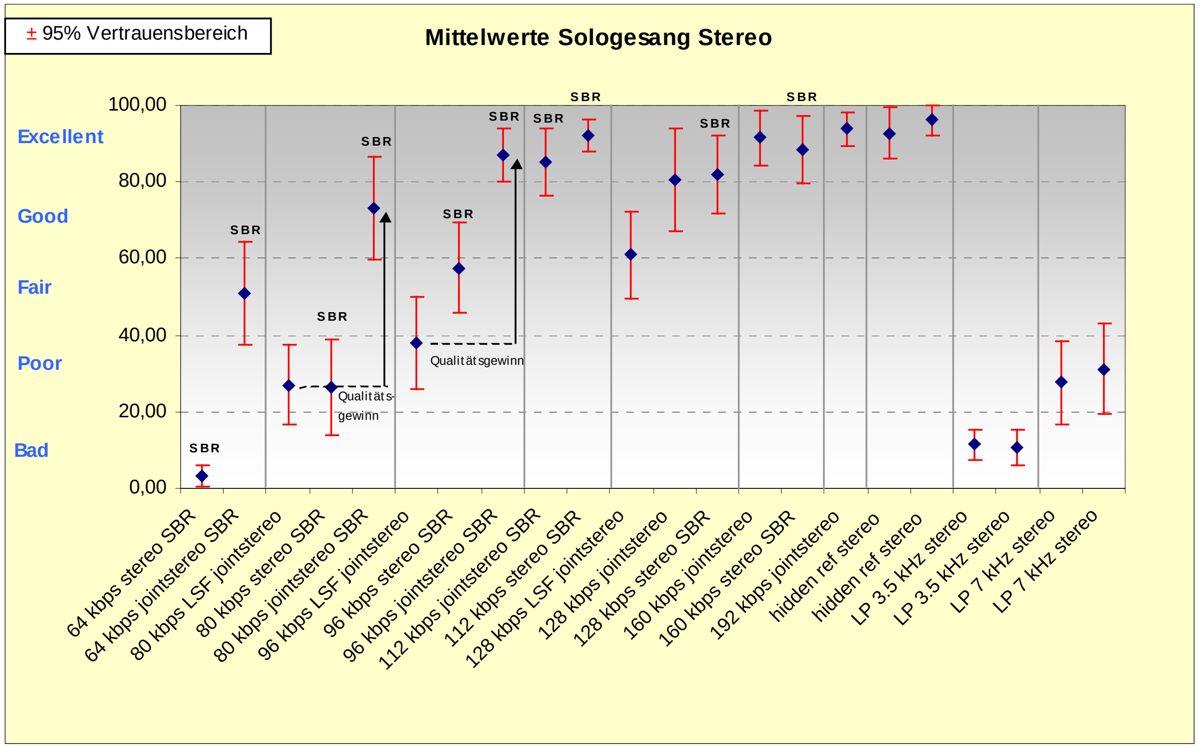
8.1.4 Stereo Pop Music



Quality scores for stereo pop music

The pop music test item shows a similar performance for stereo and joint-stereo. At 80 and 96kbit/s the differences between stereo and joint-stereo were very small with Layer II^{SBR}. However, at 64kbit/s Layer II^{SBR} the improvement joint-stereo gives over stereo provides similar performance to Layer II^{SBR} at 80kbit/s. At 128kbit/s there was an improvement in quality with standard Layer II joint-stereo and Layer II^{SBR} stereo over standard Layer II LSF joint-stereo. At 160kbit/s the coders performed similarly.

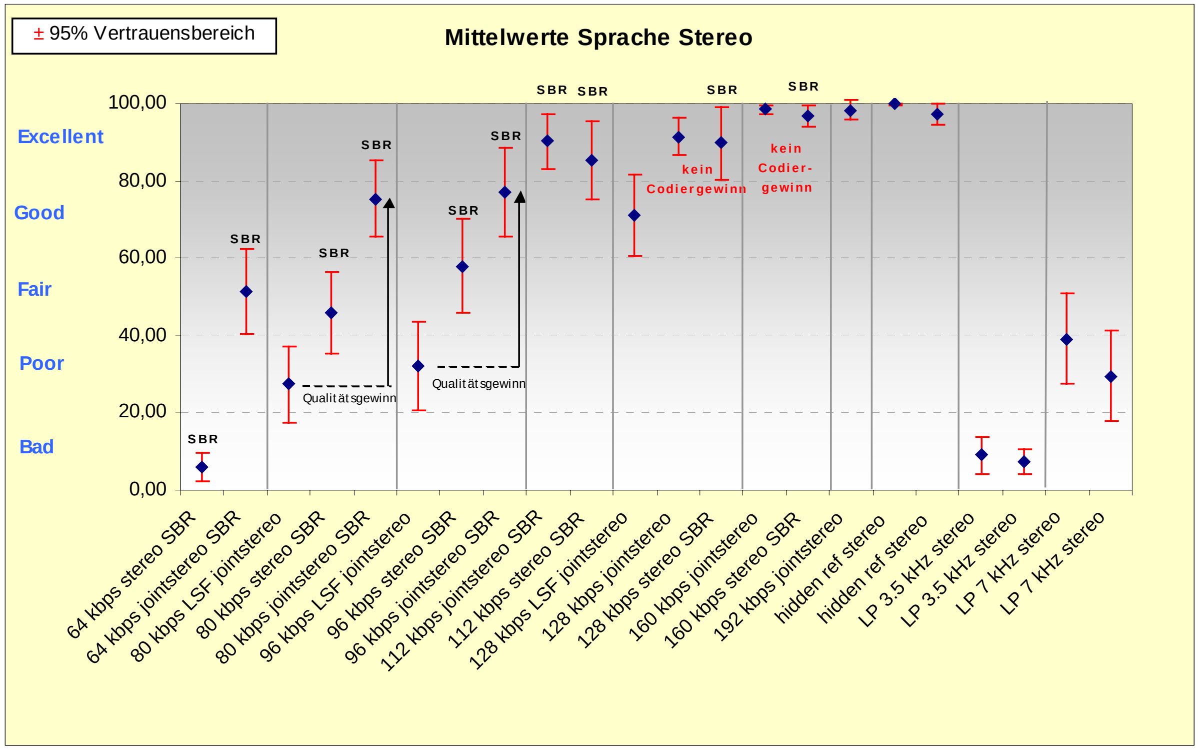
8.1.5 Stereo Solo Voice



Quality scores for stereo solo voice

The solo voice item contained monophonic female vocals with some artificial stereo ambience added. Therefore Layer II^{SBR} with joint-stereo gave the greatest increase in performance. At 128 and 160kbit/s there was no quality gain for Layer II^{SBR} over standard Layer II.

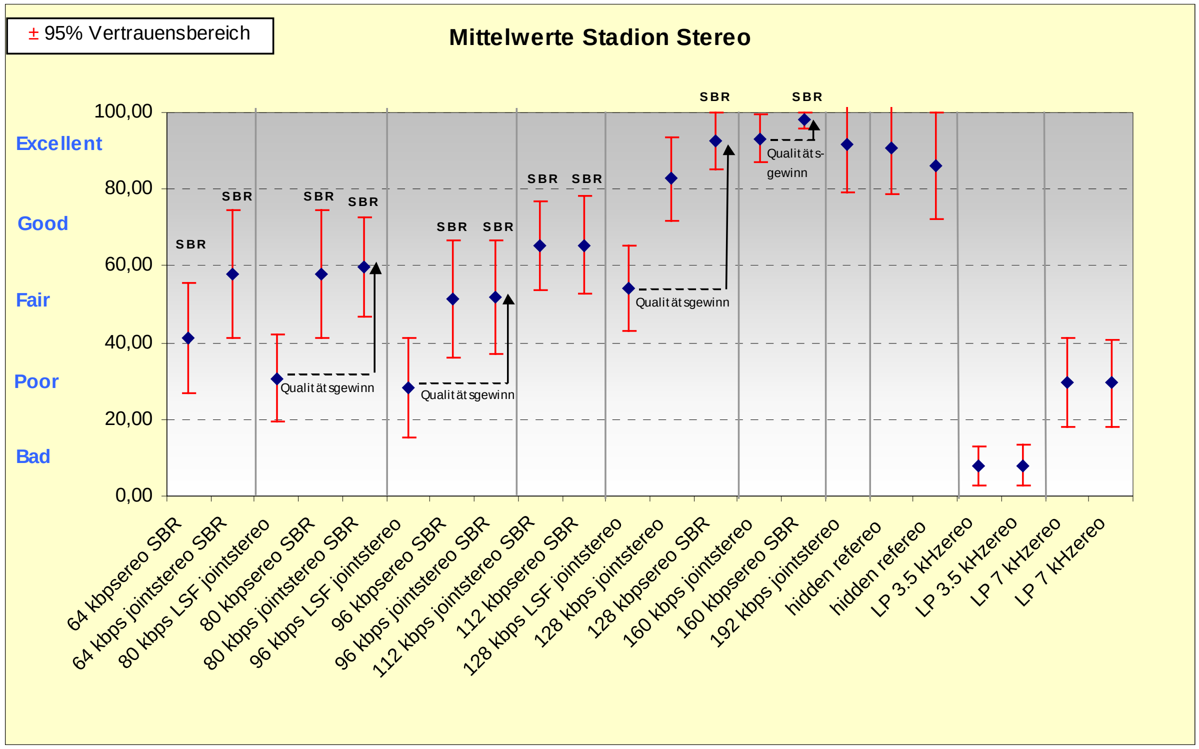
8.1.6 Stereo Speech



Quality scores for stereo speech

The speech item consisted of male speech. At lower bitrates Layer II^{SBR} gave a significant quality gain, however at 128 and 160kbit/s there was no improvement as standard Layer II joint-stereo is already classed as “excellent”.

8.1.7 Stereo Stadium

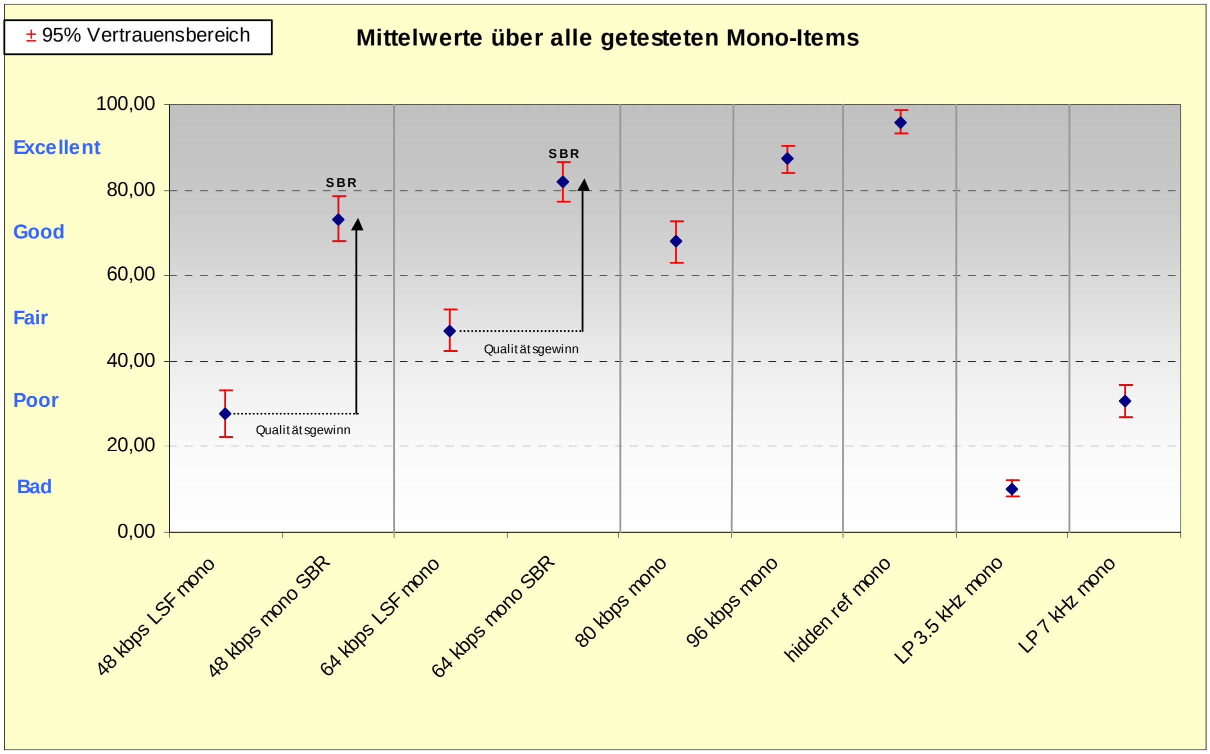


Quality scores for stereo commentary with background noise from stadium (ice hockey match)

The stadium test item contained typical background noise found in an ice hockey stadium with a female commentator speaking. The background noise consists of applause and chanting. At 128kbit/s Layer II^{SBR} was not distinguishable from the original. Only at 64kbit/s joint-stereo did Layer II^{SBR} provide some coding gain.

8.2 *Mono Test Sequence Results*

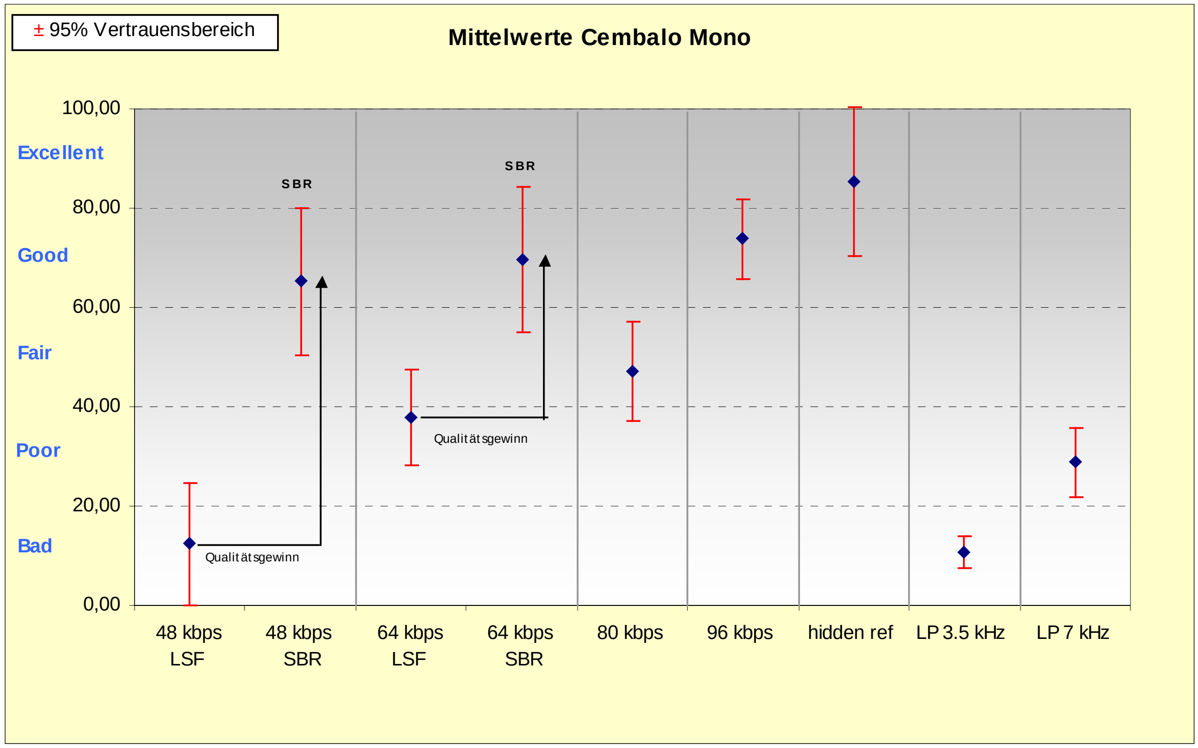
8.2.1 Mono Sequence Averages



Quality scores for mono sequences

Mono signals in broadcasting are normally used for speech-based services. Therefore only 48 and 64kbit/s were the chosen bitrates for these tests. Both 80 and 96kbit/s were used purely as additional references. At 48 and 64kbit/s Layer II coding was carried out in LSF ("half sampling rate") mode. When the stereo items are tested at 48 and 64kbit/s mono, the quality gain provided by SBR is obvious. The standard Layer II coder in LSF mode at 48kbit/s is classified as "poor", but their SBR versions are classed as "good". There is also a similar result at 64kbit/s, where the quality increases from "fair" to "excellent". At both these bitrates the SBR quality is higher than standard Layer II running at 80kbit/s mono.

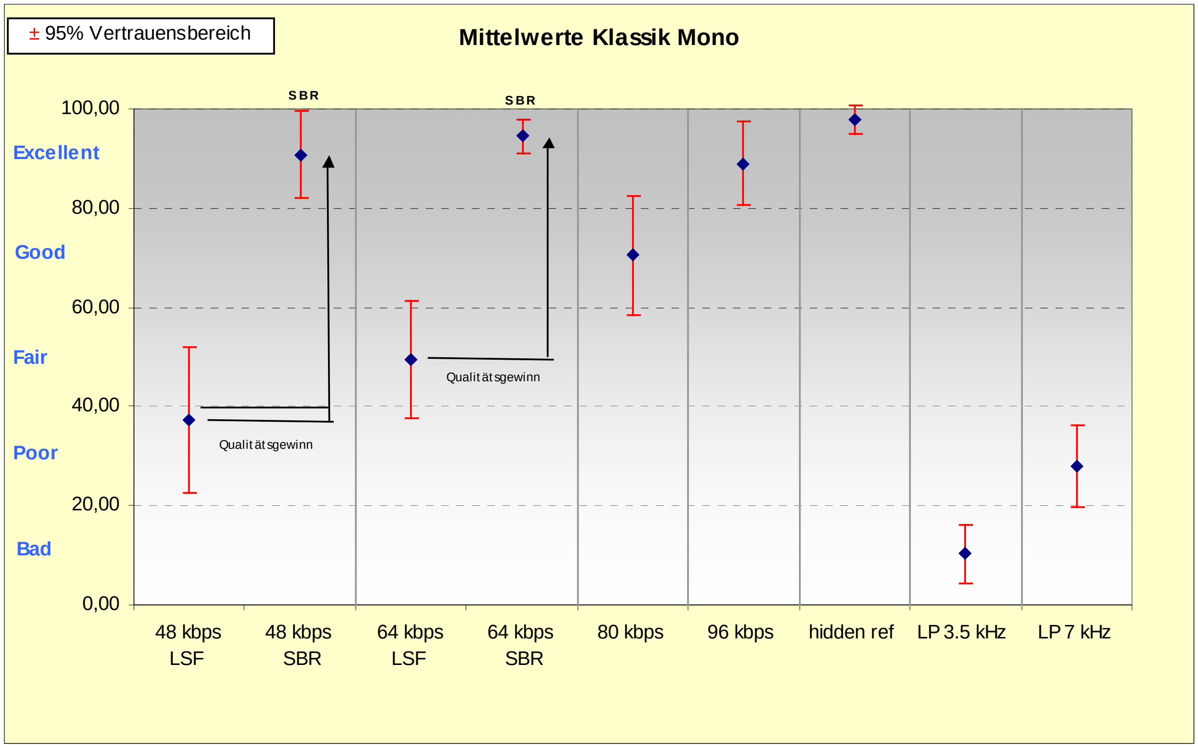
8.2.2 Mono Harpsichord



Quality scores for mono harpsichord

The harpsichord was found to be the most critical item in the stereo tests, and the same pattern is shown here for the mono tests, where the resulting quality gain is lower. With Layer II^{SBR} coding both 48 and 64kbit/s reach the classification of "good". For the standard Layer II it requires 96kbit/s to reach this classification.

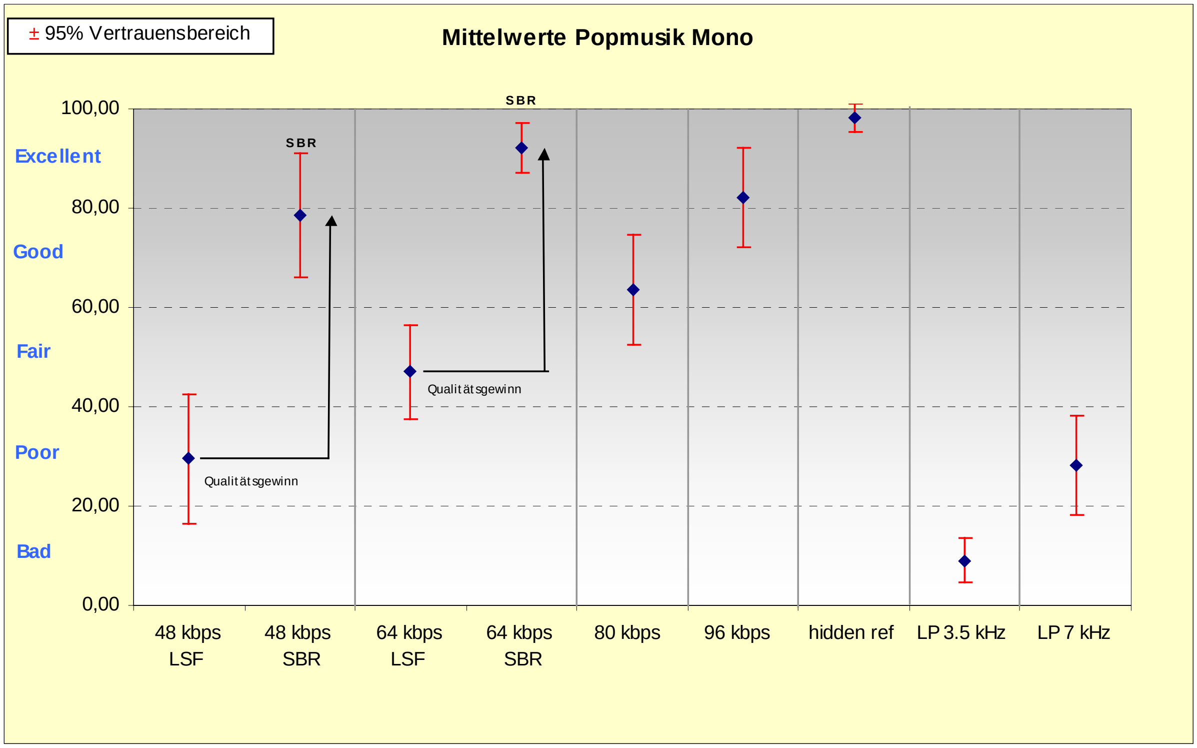
8.2.3 Mono Classical



Quality scores for mono classical music

Much like the stereo tests for classical music, the mono tests shows are similar improvement for Layer II^{SBR} over standard Layer II. Layer II^{SBR} getting close to the hidden reference at both 48 and 64kbit/s.

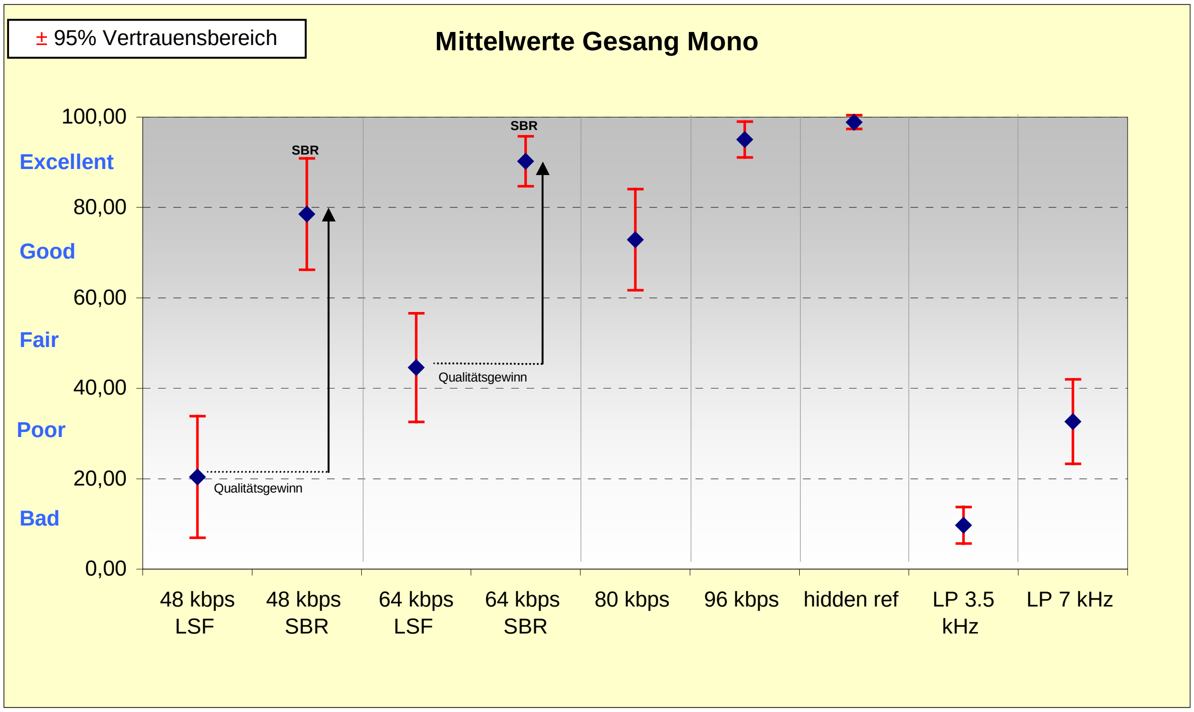
8.2.4 Mono Pop Music



Quality scores for mono pop music

The pop music item coded at low bitrates with Layer II^{SBR} gives a large improvement in audio quality. Even at 48kbit/s the quality classification jumps from “poor” (with standard Layer II) to “good”; and at 64kbit/s from “fair” to “excellent”. At 48kbit/s Layer II^{SBR} is comparable to standard Layer II at 96kbit/s, with 64kbit/s with Layer II^{SBR} better still. The hidden reference was clearly identified by all the participants.

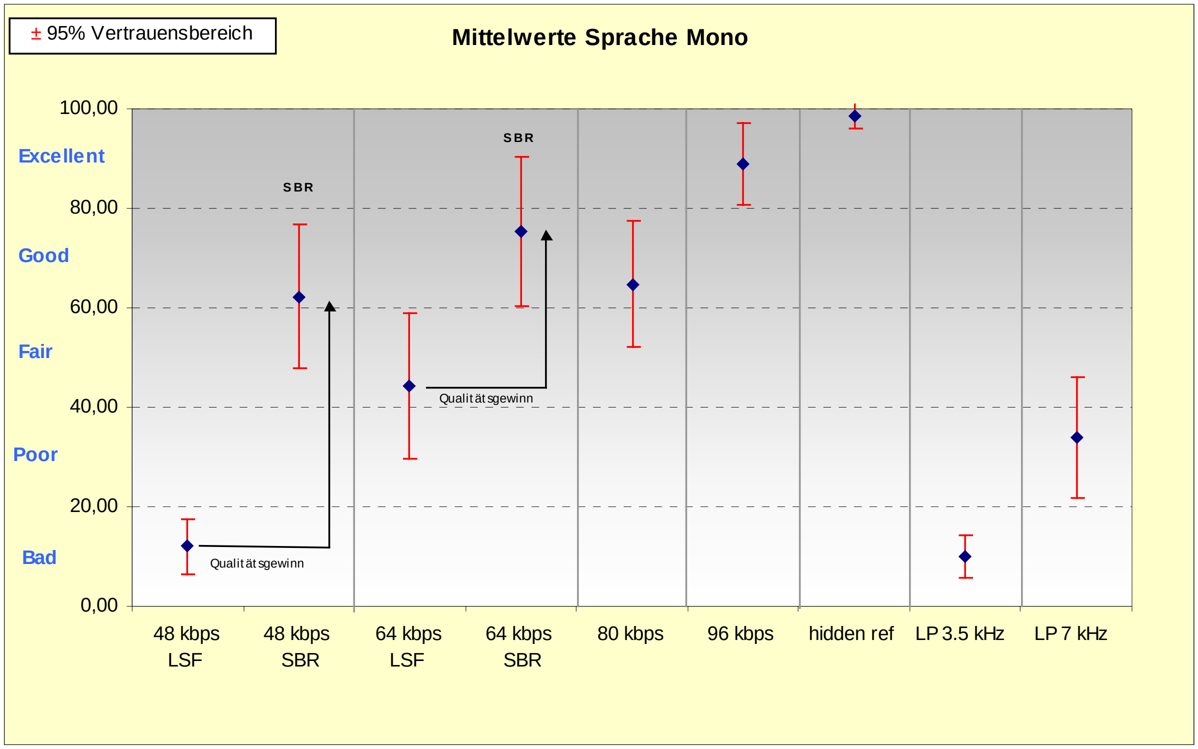
8.2.5 Mono Solo Voice



Quality scores for mono solo voice

The solo voice test items produced similar results to the pop music item with a slightly higher quality gain.

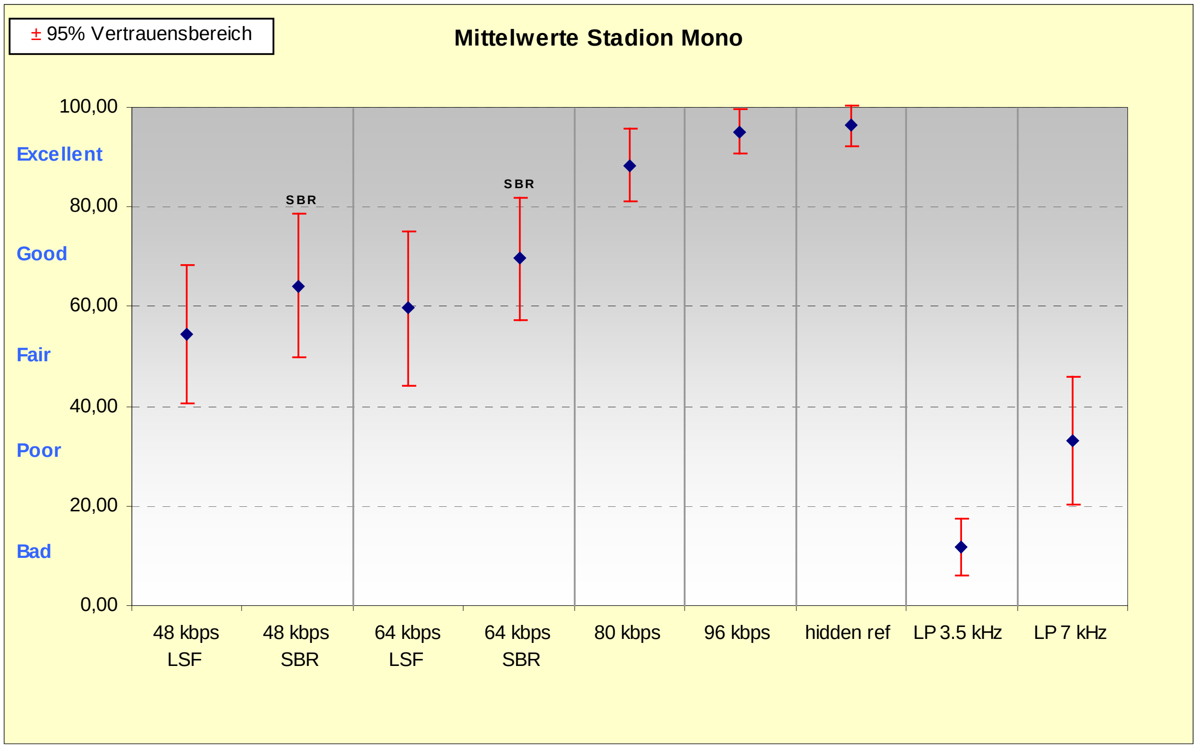
8.2.6 Mono Speech



Quality scores for mono speech

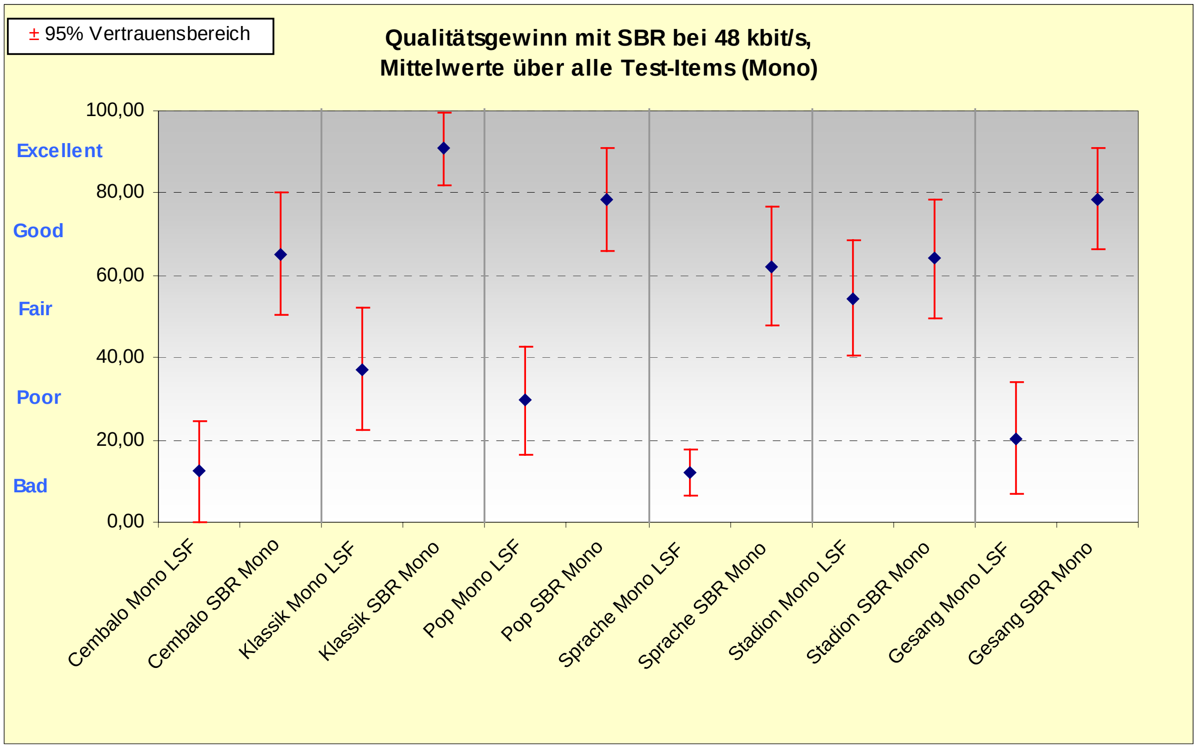
The male speech item produced similar results to the harpsichord item, and the hidden reference was clearly identified by all the participants.

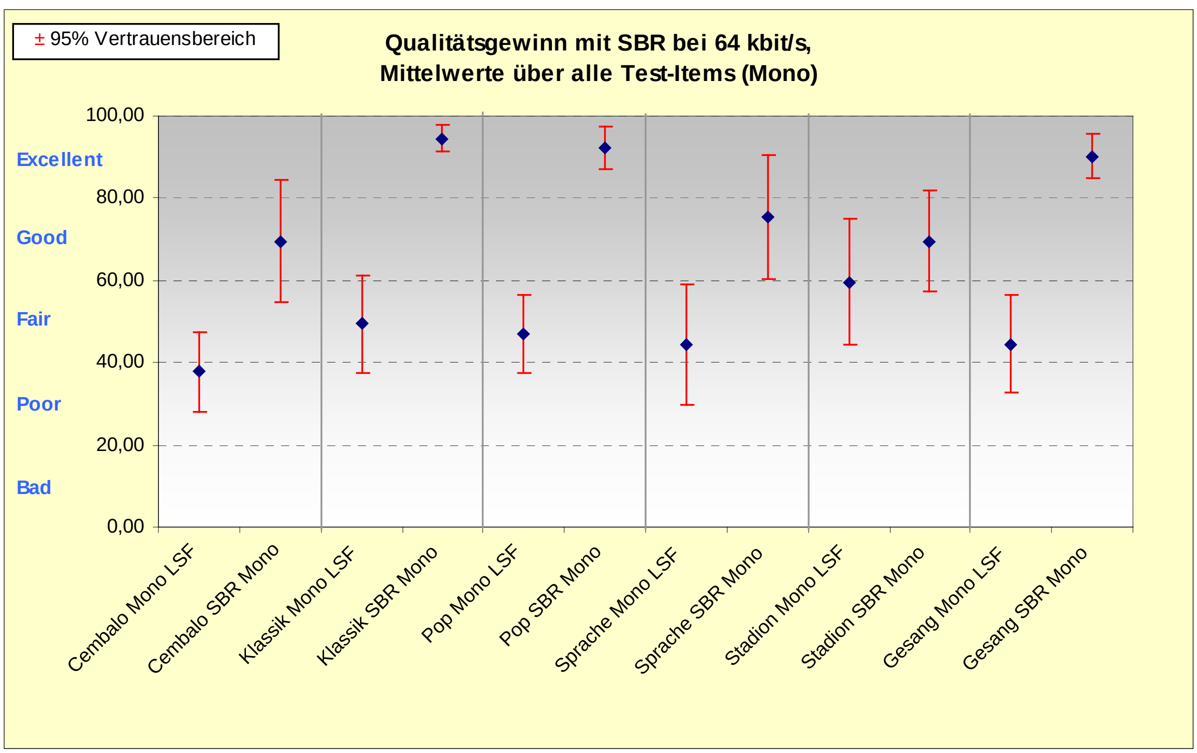
8.2.7 Mono Stadium

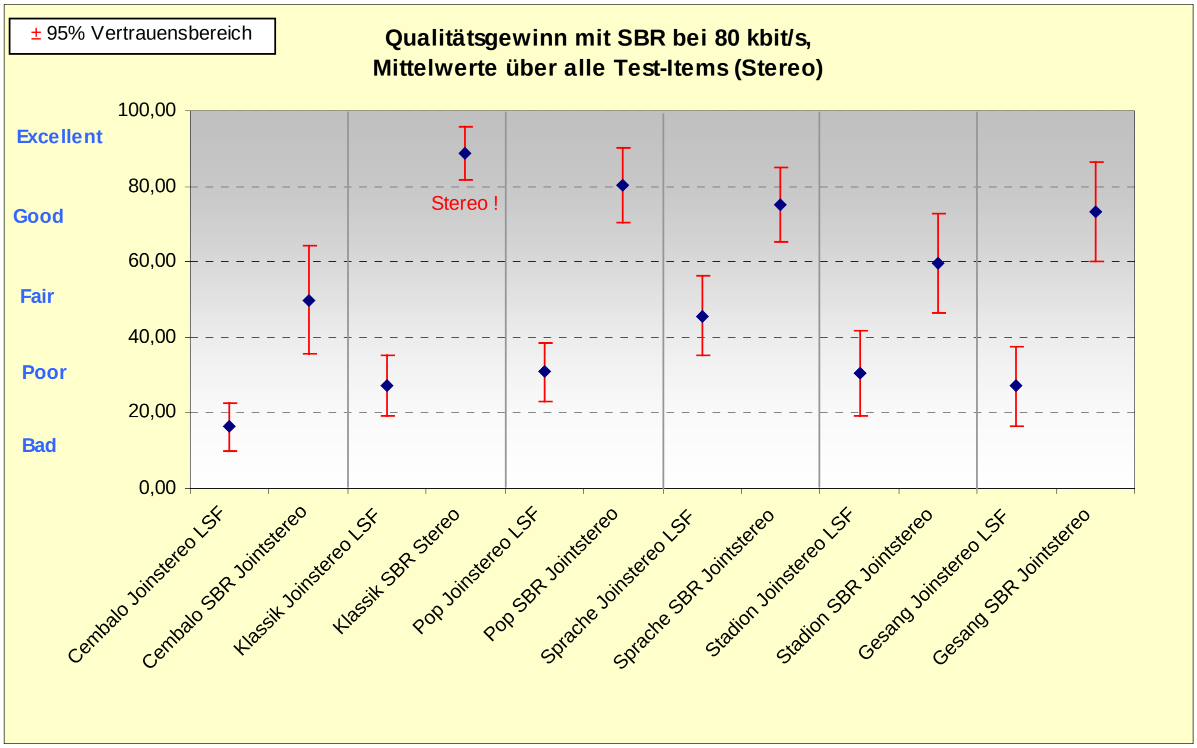


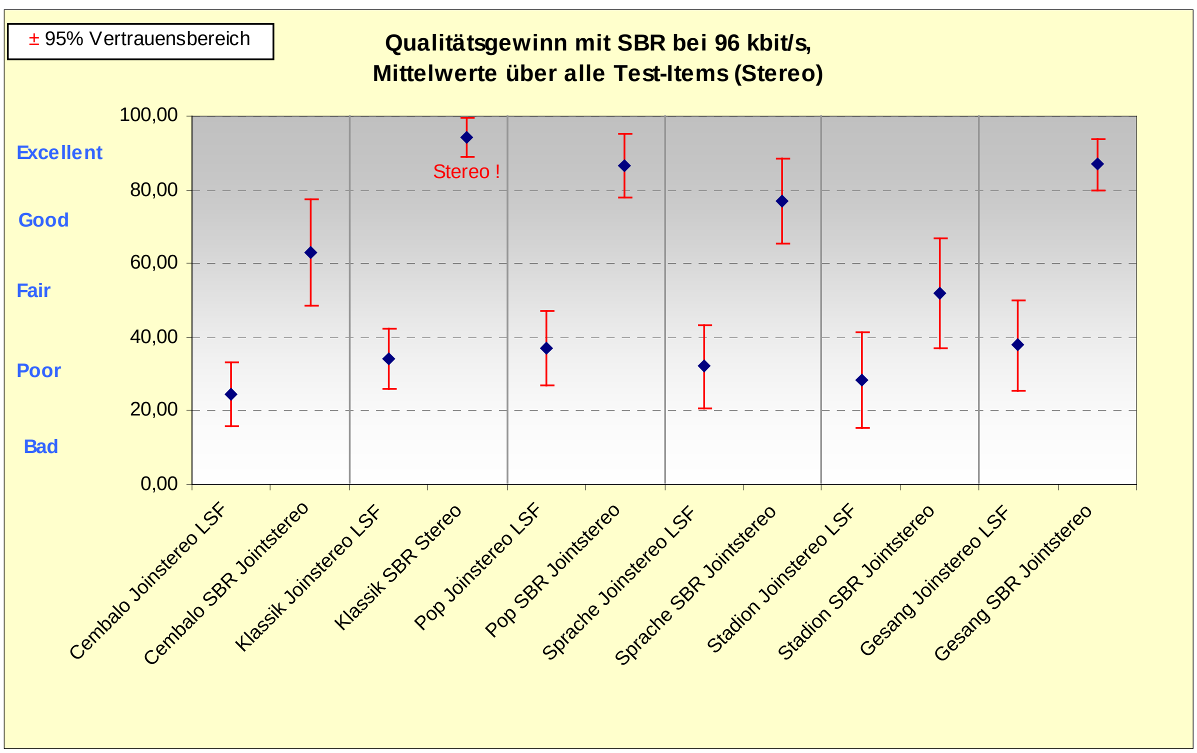
The stadium test item achieves high quality scores even with the standard Layer II codec. The original signal contains very few frequencies above 9kHz, which explains why SBR offers no significant improvements. Despite the limited bandwidth of the item, the hidden reference was clearly identified by all the participants.

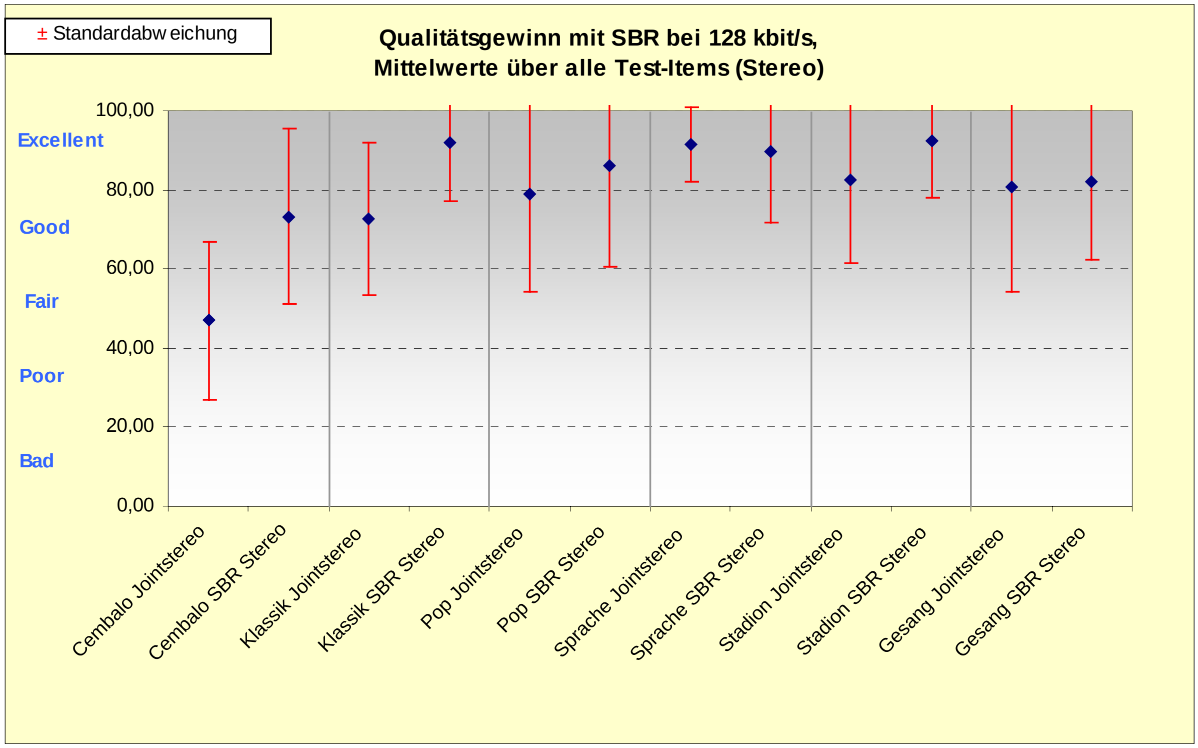
8.3 Quality gain scores for each bitrate

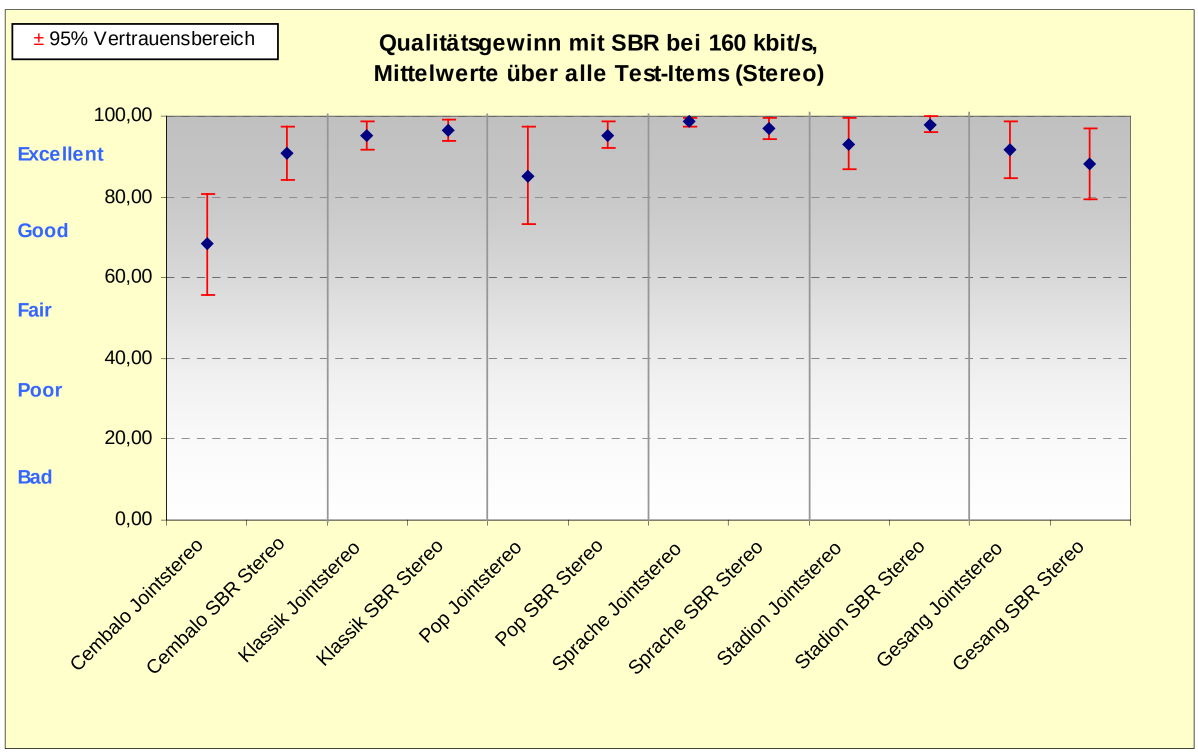




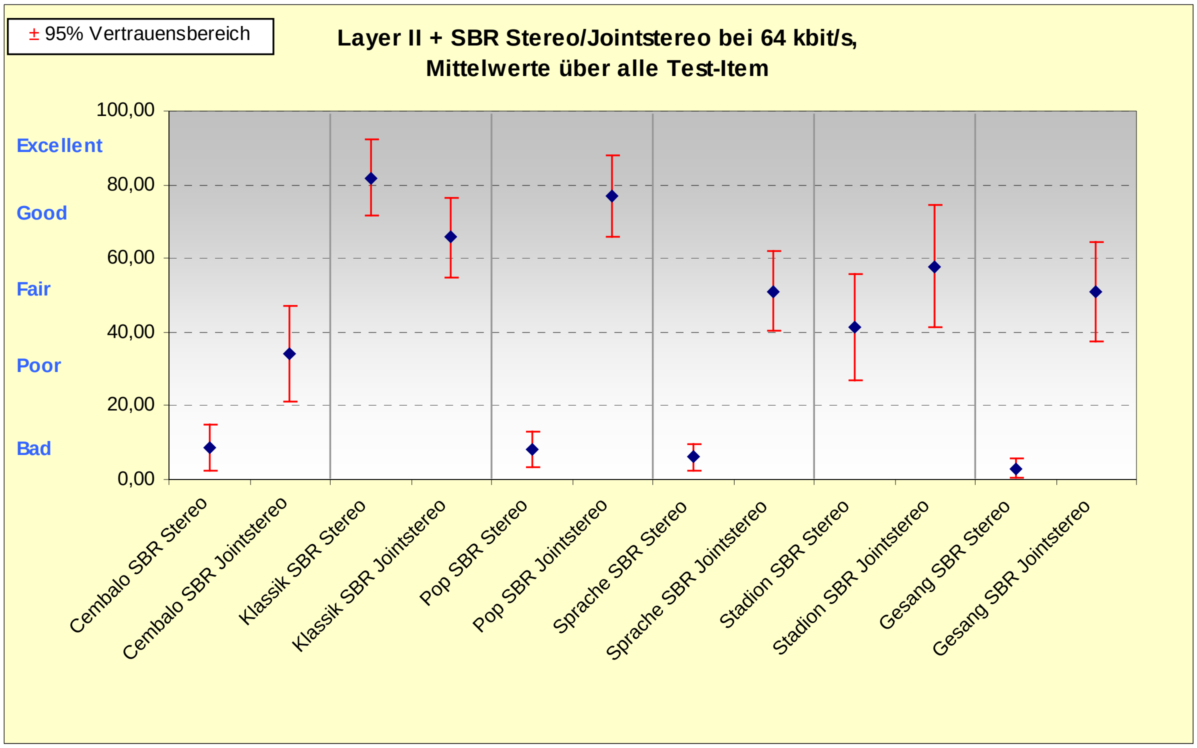

± 95% Vertrauensbereich
Qualitätsgewinn mit SBR bei 80 kbit/s,
Mittelwerte über alle Test-Items (Stereo)
Excellent
Good
Fair
Poor
Bad
100,00
80,00
60,00
40,00
20,00
0,00
Stereo !
Cembalo Joinstereo LSF
Cembalo SBR Jointstereo
Klassik Joinstereo LSF
Klassik SBR Stereo
Pop Joinstereo LSF
Pop SBR Jointstereo
Sprache Joinstereo LSF
Sprache SBR Jointstereo
Stadion Joinstereo LSF
Stadion SBR Jointstereo
Gesang Joinstereo LSF
Gesang SBR Jointstereo

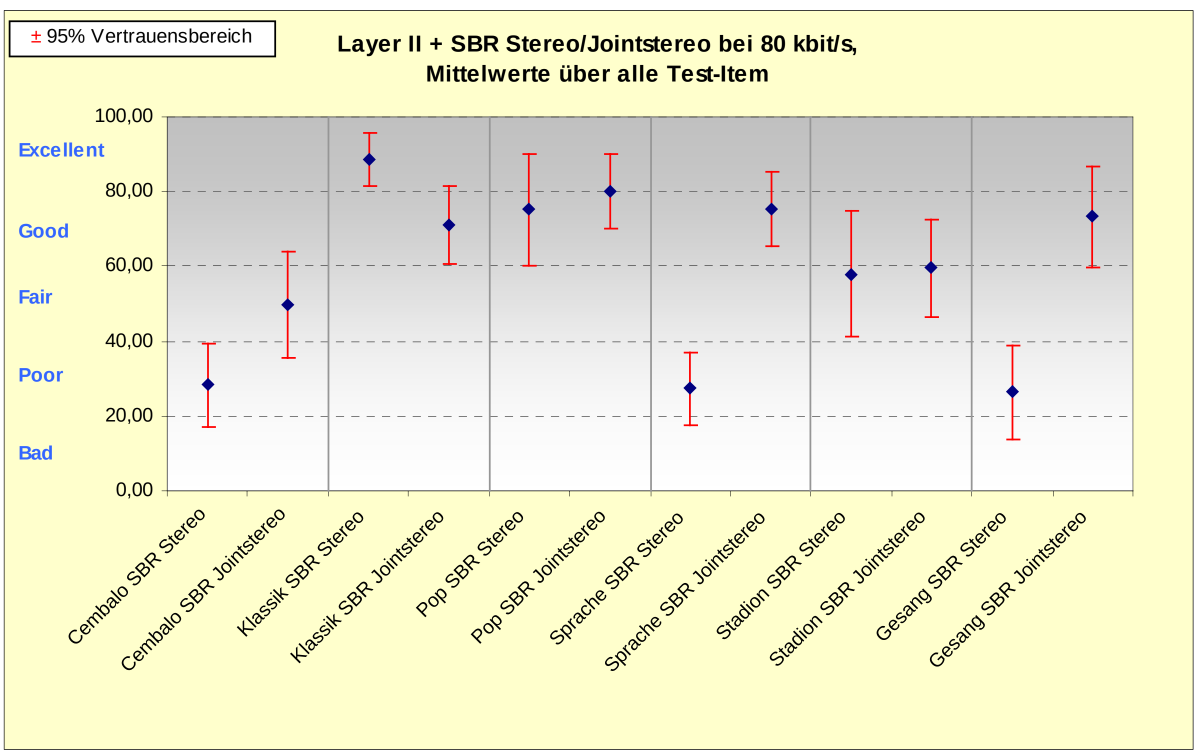

± 95% Vertrauensbereich
Qualitätsgewinn mit SBR bei 96 kbit/s,
Mittelwerte über alle Test-Items (Stereo)
Excellent
Good
Fair
Poor
Bad
100,00
80,00
60,00
40,00
20,00
0,00
Stereo !
Cembalo Joinstereo LSF
Cembalo SBR Jointstereo
Klassik Joinstereo LSF
Klassik SBR Stereo
Pop Joinstereo LSF
Pop SBR Jointstereo
Sprache Joinstereo LSF
Sprache SBR Jointstereo
Stadion Joinstereo LSF
Stadion SBR Jointstereo
Gesang Joinstereo LSF
Gesang SBR Jointstereo

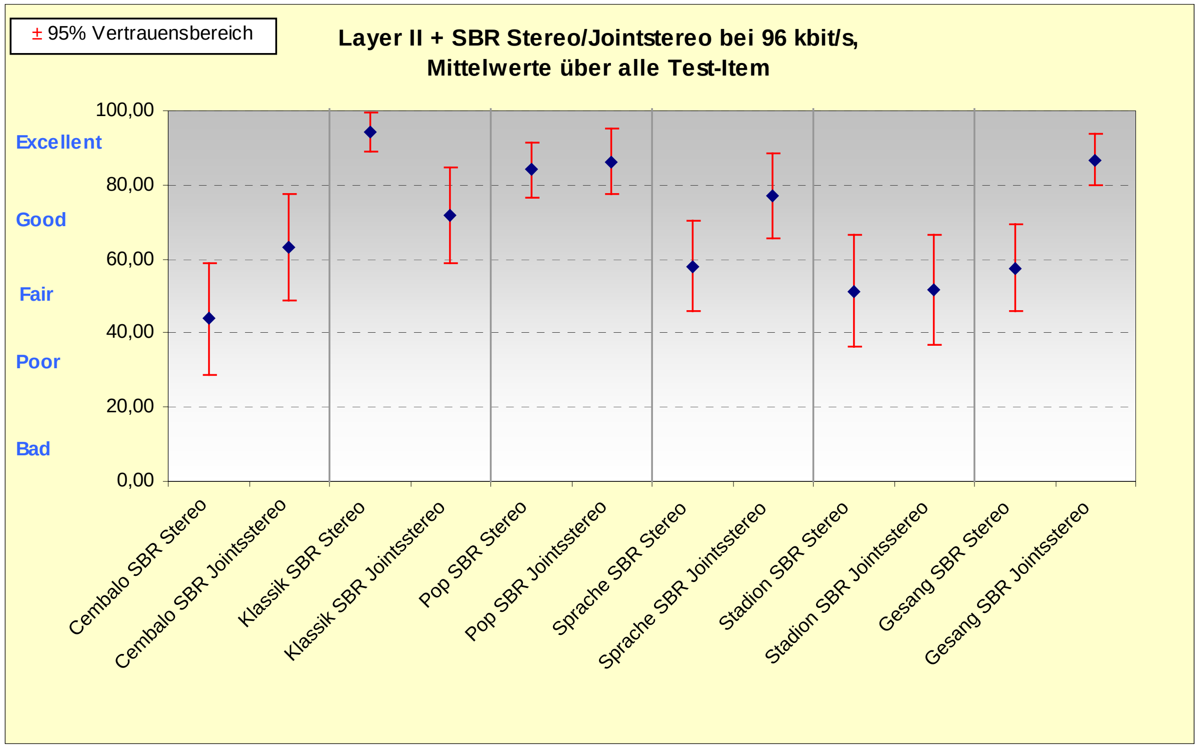

± Standardabweichung
Qualitätsgewinn mit SBR bei 128 kbit/s,
Mittelwerte über alle Test-Items (Stereo)
100,00
80,00
60,00
40,00
20,00
0,00
Excellent
Good
Fair
Poor
Bad
Cembalo Jointstereo
Cembalo SBR Stereo
Klassik Jointstereo
Klassik SBR Stereo
Pop Jointstereo
Pop SBR Stereo
Sprache Jointstereo
Sprache SBR Stereo
Stadion Jointstereo
Stadion SBR Stereo
Gesang Jointstereo
Gesang SBR Stereo

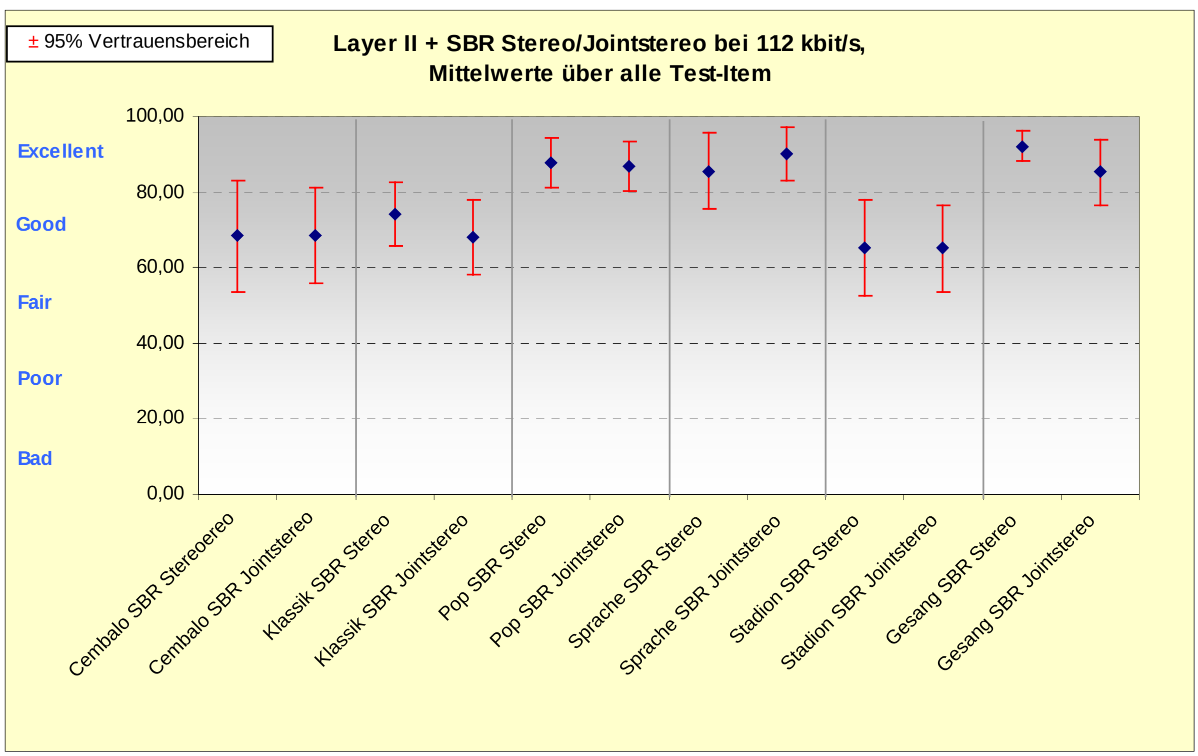

± 95% Vertrauensbereich
Qualitätsgewinn mit SBR bei 160 kbit/s,
Mittelwerte über alle Test-Items (Stereo)
100,00
80,00
60,00
40,00
20,00
0,00
Excellent
Good
Fair
Poor
Bad
Cembalo Jointstereo
Cembalo SBR Stereo
Klassik Jointstereo
Klassik SBR Stereo
Pop Jointstereo
Pop SBR Stereo
Sprache Jointstereo
Sprache SBR Stereo
Stadion Jointstereo
Stadion SBR Stereo
Gesang Jointstereo
Gesang SBR Stereo

8.4 Results Quality Gain sorted according to Stereo-/Joint Stereo-Mode






± 95% Vertrauensbereich
Layer II + SBR Stereo/Jointstereo bei 96 kbit/s,
Mittelwerte über alle Test-Item
100,00
80,00
60,00
40,00
20,00
0,00
Excellent
Good
Fair
Poor
Bad
Cembalo SBR Stereo
Cembalo SBR Jointsstereo
Klassik SBR Stereo
Klassik SBR Jointsstereo
Pop SBR Stereo
Pop SBR Jointsstereo
Sprache SBR Stereo
Sprache SBR Jointsstereo
Stadion SBR Stereo
Stadion SBR Jointsstereo
Gesang SBR Stereo
Gesang SBR Jointsstereo


± 95% Vertrauensbereich
Layer II + SBR Stereo/Jointstereo bei 112 kbit/s,
Mittelwerte über alle Test-Item
100,00
80,00
60,00
40,00
20,00
0,00
Excellent
Good
Fair
Poor
Bad
Cembalo SBR Stereoereo
Cembalo SBR Jointstereo
Klassik SBR Stereo
Klassik SBR Jointstereo
Pop SBR Stereo
Pop SBR Jointstereo
Sprache SBR Stereo
Sprache SBR Jointstereo
Stadion SBR Stereo
Stadion SBR Jointstereo
Gesang SBR Stereo
Gesang SBR Jointstereo

9 Coding Gain of SBR

The **coding gain** achievable by upgrading from standard Layer II to Layer II^{SBR} can be deduced from the **quality gain** (results in section 8.1). The **coding gain** can be defined as the reduction in bitrate obtained by Layer II^{SBR} to produce the same audio quality as standard Layer II. The **coding gain** can be estimated from the average value of all the test items and sequences using this formula:

$$\text{Coding Gain } g = \left(\frac{B_{\text{norm.Layer II}} - B_{\text{Layer II SBR}}}{B_{\text{norm.Layer II}}} \right) \cdot 100\,\% \text{ (at equal audio quality)}$$

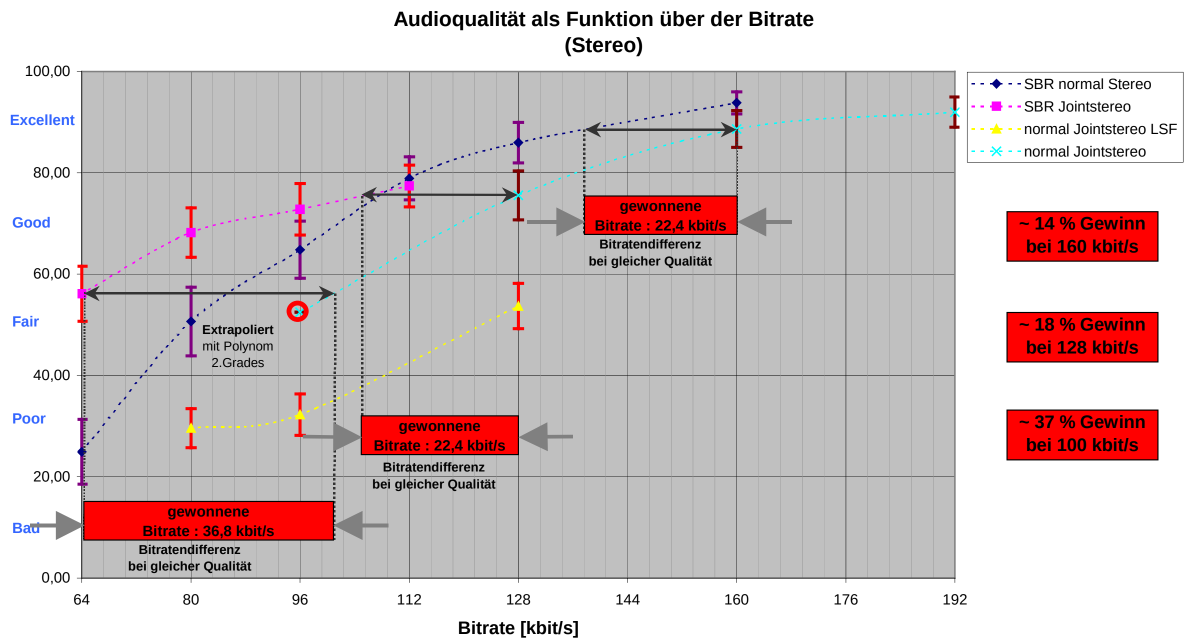
whereby B = Bit rate and g = coding Gain in %

Equation 1

9.1 Coding Gain for stereo sequences

The following diagram shows how audio quality is estimated as a function of bitrate (intermediate values determined with interpolation, 95% confidence interval). The four coding methods which are compared are:

- Layer II^{SBR} stereo (blue)
- Layer II^{SBR} joint-stereo (purple)
- Standard Layer II joint-stereo (light-blue)
- Standard Layer II joint-stereo LSF (yellow)



All the coding methods have an increasing audio quality with respect to bitrate, but level off to a maximum value. Above approximately 109kbit/s Layer II^{SBR} joint-stereo will not give any benefit over Layer II^{SBR} stereo (compare magenta line with blue line). Therefore it is reasonable to switch into full stereo mode above 112kbit/s with Layer II^{SBR}.

Standard Layer II joint-stereo (light blue line) and Standard Layer II LSF (yellow line) display an inferior audio quality compared to Layer II^{SBR}. However, it must be noted that the unexpectedly poor quality of Layer II LSF at low bitrates contradicts an earlier test recommendation [6] for LSF coding below 128kbit/s. Standard Layer II joint-stereo (light blue line) was determined from an extrapolation process.

The figure above the **coding gain** is illustrated with three selected bitrates. From this we can determine what the average coding gain can be expected when upgrading from standard Layer II to Layer II^{SBR} at the following bitrates:

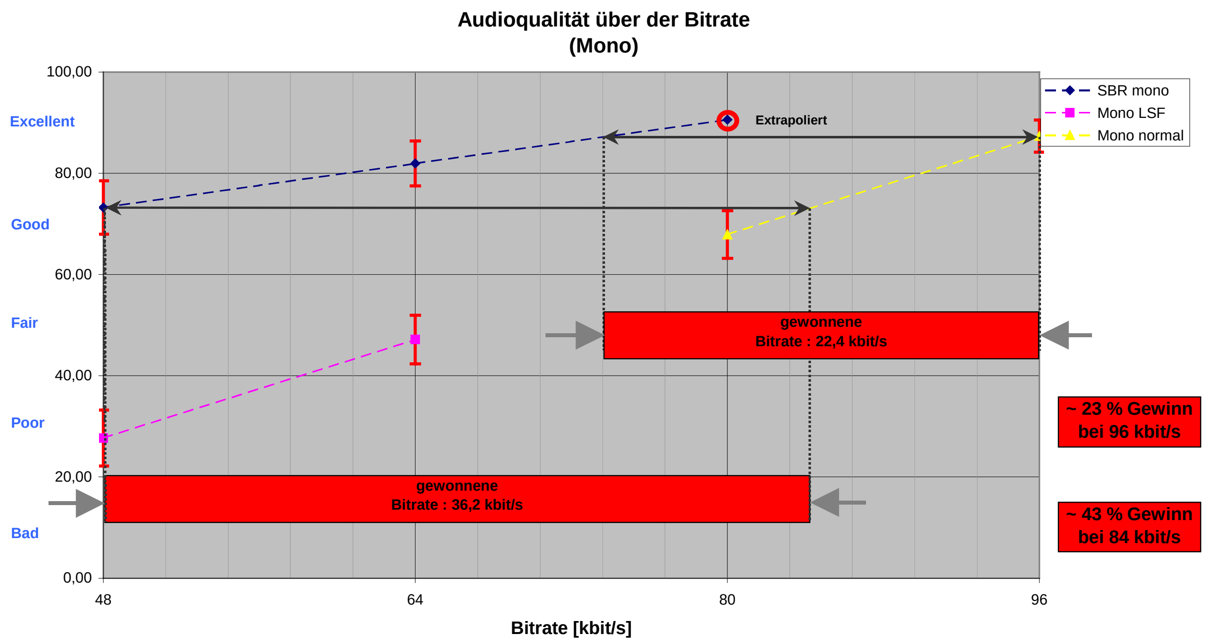
Bit-rate of standard Layer II joint-stereo	Coding gain (bit rate reduction) with SBR
160 kbit / s	14 %
128 kbit / s	18 %
~ 100 kbit / s	37 %

9.2 Coding Gain for Mono Sequences

The results found with the mono test items were similar to that of the stereo items.

- Layer II^{SBR} mono (blue)
- Standard Layer II mono LSF (magenta)
- Standard Layer II mono (yellow)

95% confidence intervals are also included.



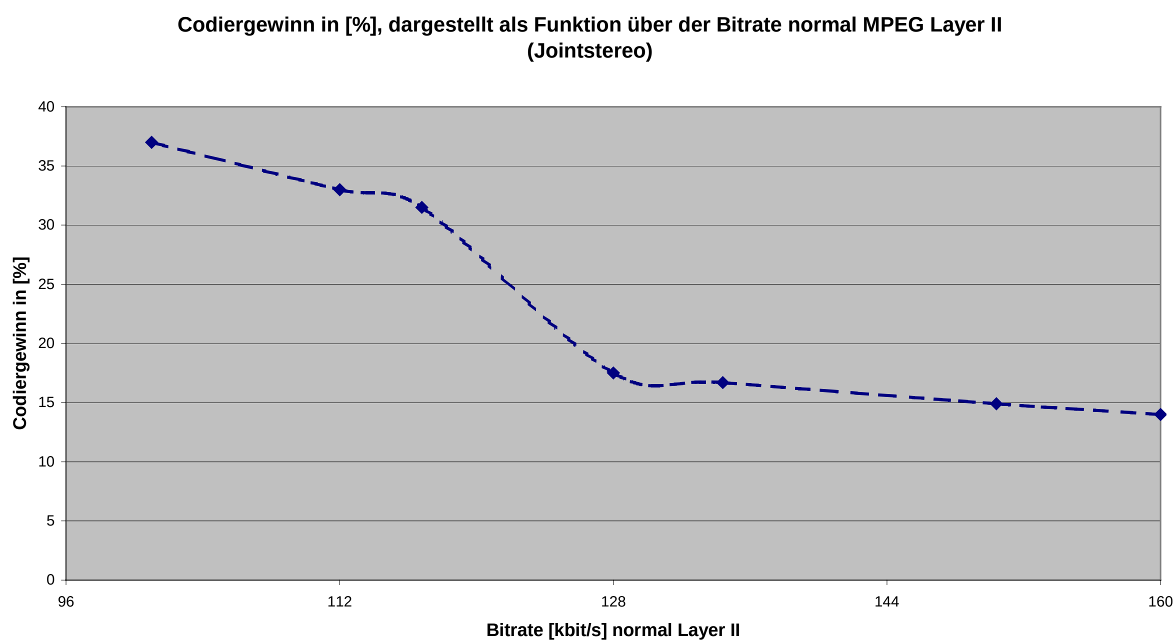
Only two test items were available for each coding method. The trend between the two measurements was obtained by interpolation, and the Layer II^{SBR} curve was determined by extrapolation.

Bit-rate of standard Layer II mono	Coding gain (bit rate reduction) with SBR
84 kbit / s	43 %
96 kbit / s	23 %

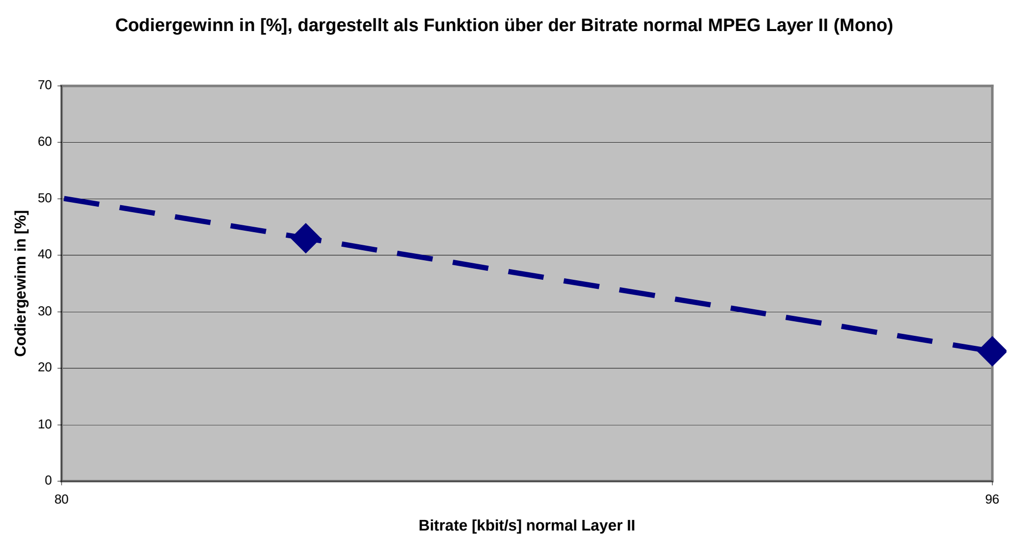
9.3 *Coding Gain (Layer II^{SBR}) as a function of bitrate*

The relationship between the coding gain achieved with using SBR against bitrate of standard Layer II is shown in the following figures. The curves show the amount of bitrate that can be saved without degrading the audio quality by upgrading from standard Layer II to Layer II^{SBR}.

Stereo



Mono



In both mono and stereo modes the large advantage of SBR at low to middle bitrates is clear. However, at higher bitrates standard Layer II gives sufficient quality so the coding gain of SBR is less. Very low bitrates were not used in this study.

9.4 Layer II^{SBR} bit rate as a function of standard Layer II bitrate

Using equation 1 above the resulting bitrate of Layer II^{SBR} can be calculated:

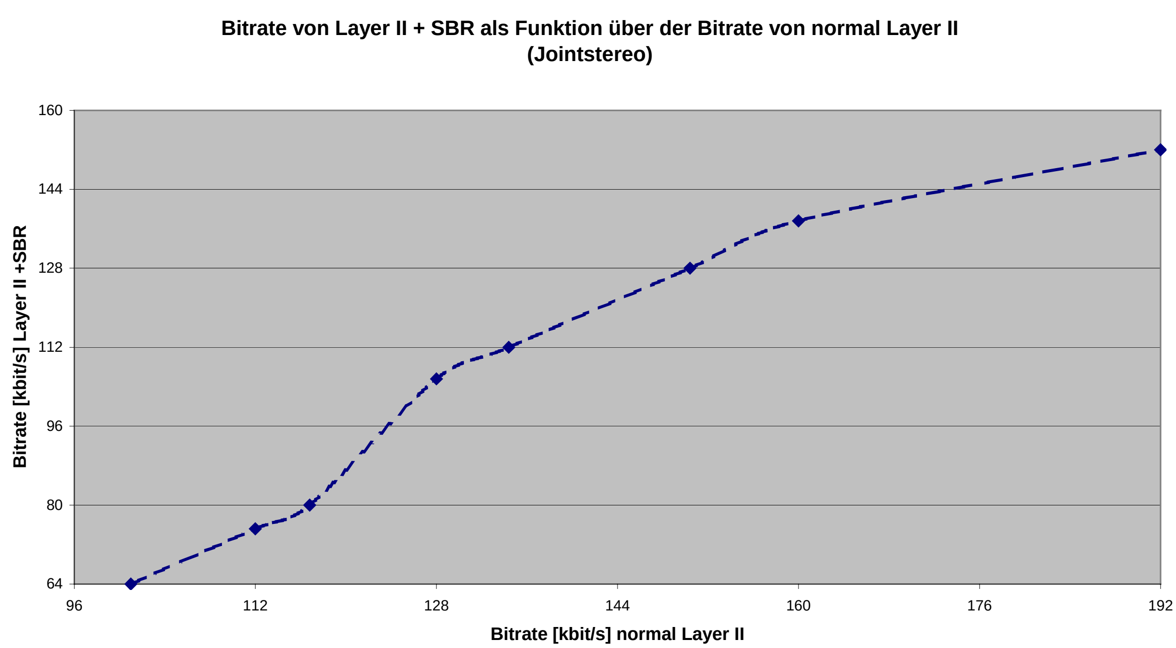
$$B_{\text{Layer II SBR}} = B_{\text{norm.Layer II}} - \left(\frac{g \cdot B_{\text{norm.Layer II}}}{100} \right)$$

where B = Bit rate and g = coding Gain in %

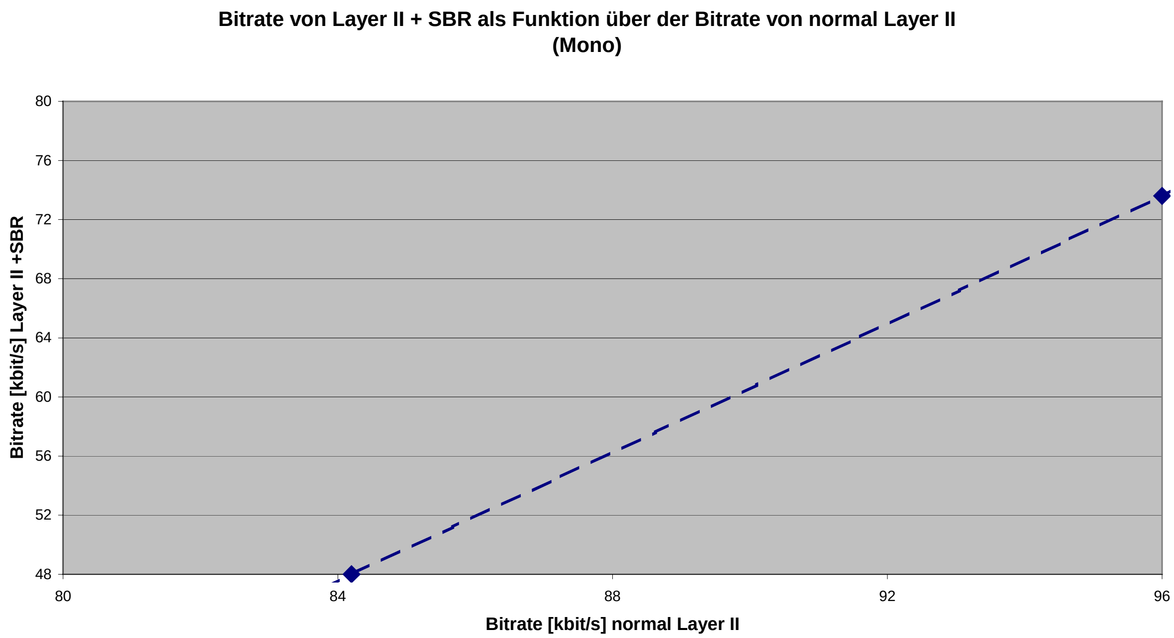
Equation 2

The bitrates of standard Layer II and Layer II^{SBR} can be compared to show the bitrate saving possible. This is shown for both stereo and mono modes in the following figures.

Stereo



Mono

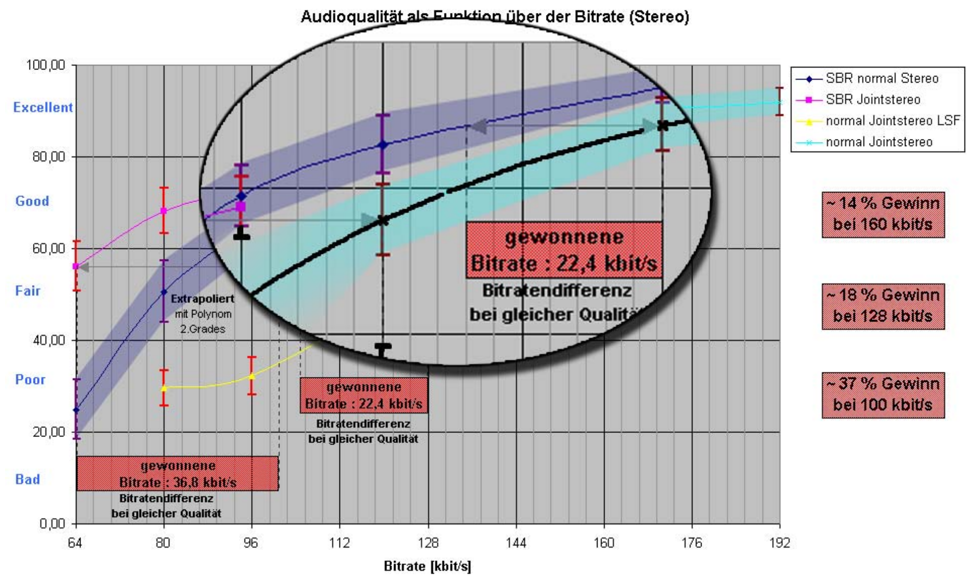


9.5 *Uncertainty of the Analysis*

A study was carried out to estimate the worst possible coding gain. As the results varied greatly across the test items, a worse case item was chosen (coding gain of 14% at 160kbit/s Layer II joint-stereo) which had the widest confidence intervals.

The optimistic answer for 160kbit/s gave a saving of about 45kbit/s (coding gain of 28%), whereas the pessimistic answer gave no coding gain.

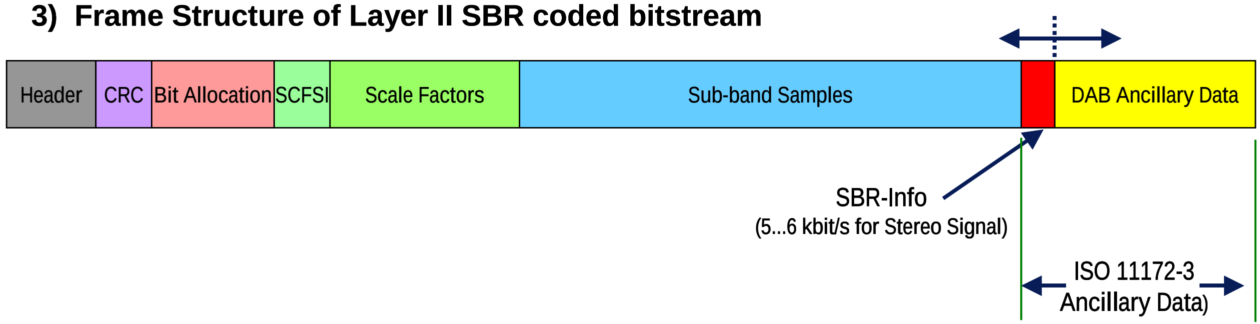
To try and minimise the variance of the result, only trained listeners were considered, as they should be more critical in their scoring.



10 Quality degradation

If SBR-encoded audio signals are broadcast and decoded by a normal MPEG Layer 2 decoder, some quality penalty is to be paid due to lower usable bitrate (namely, some bits are taken by the SBR process). The insertion of the SBR Information („Payload") into the DAB data stream results in a reduction of available bitrate for coding of the basic data stream by 5 to 6 kbit/s. If a normal Layer 2 decoder is used to decode the SBR coded audio, the introduction of SBR may lead to some degradation of audio quality, especially at lower bitrates.

3) Frame Structure of Layer II SBR coded bitstream



In order to assess whether this penalty is significant or not, several tests were conducted by IRT. The tables below show all the combinations taken. The following bitrates were used:

Stereo mode: 160 / 128 / 112 / 96 kbit/s

Mono mode: 80 / 64 / 48 kbit/s.

For detailed quality degradation results see **Appendix 2**. This Appendix provides our findings for the different audio items:

- harpsichord
- male speech
- stadium noise
- classical music
- pop music
- female voice.

Stereo test

	CASE 1	CASE 2	CASE 3
Encoder	SBR-encoded audio items	SBR-encoded audio items	Normal L2 encoded items
Decoder	SBR-decoded audio items	Normal L2 decoded items	Normal L2 decoded items
Data rate Mode	A) 160 kbit/s Joint, Single rate B) 128 kbit/s Joint, Single rate C) 112 kbit/s Joint, Single rate D) 96 kbit/s Joint, Dual-rate	A)160 kbit/s Joint, Single-rate B)128 kbit/s Joint, Single-rate C) 112 kbit/s Joint, Single-rate D) 96 kbit/s Joint, Dual-rate	96 kbit/s Joint, Half-Sampling-rate

The tests were divided into two sessions: first session with stereo signals, second session with mono signals.

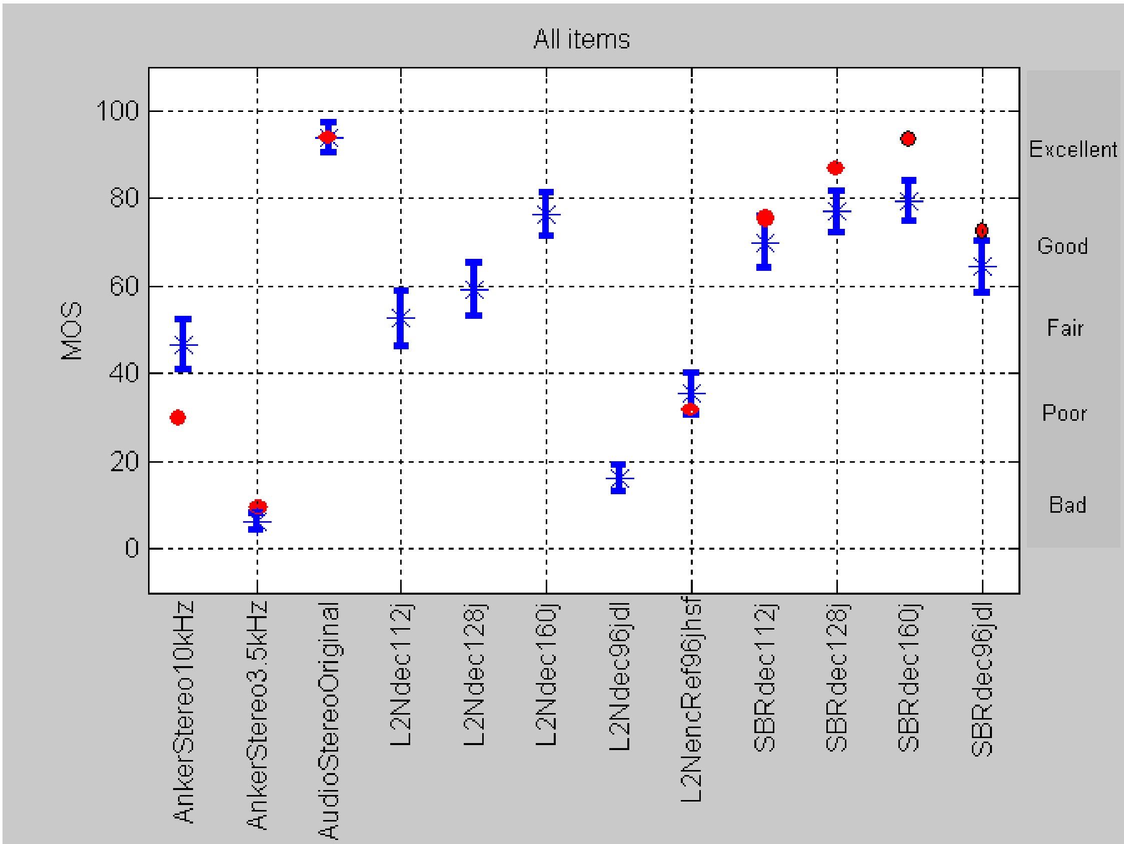
The MUSHRA methodology was used (hidden reference, anchors: 3.5 and 10kHz).

Mono test

	CASE 1	CASE 2	CASE 3
Encoder	SBR-encoded audio items	SBR-encoded audio items	Normal L2 encoded items
Decoder	SBR-decoded audio items	Normal L2 decoded items	Normal L2 decoded items
Data-rate Mode	A) 80 kbit/s Mono, Single rate B) 64 kbit/s Mono, Dual rate C) 48 kbit/s Mono, Dual rate	A) 80 kbit/s Mono, Single rate B) 64 kbit/s Mono, Dual rate C) 48 kbit/s Mono, Dual rate	64 kbit/s Mono, Half-Sampling-rate

10.1 Stereo Test Results

With 12 test subjects, the following results were obtained:



Legend:

AnkerStereo10kHz	Stereo anchor 10 kHz
AnkerStereo3.5kHz	Stereo anchor 3.5 kHz
AnkerStereoOriginal	Reference (hidden)
L2Ndec112j	SBR-encoded, L2-decoded, joint-stereo, single rate, 112 kbit/s
L2Ndec128j	SBR-encoded, L2-decoded, joint-stereo, single rate, 128 kbit/s
L2Ndec160j	SBR-encoded, L2-decoded, joint-stereo, single rate, 160 kbit/s
L2Ndec96jdl	SBR-encoded, L2-decoded, joint-stereo, dual rate, 96 kbit/s
L2NencRef96jhsf	L2-encoded, L2-decoded, joint stereo, half-sampling rate, 96 kbit/s
SBRdec112j	SBR-encoded, SBR-decoded, joint stereo, single rate, 112 kbit/s
SBRdec128j	SBR-encoded, SBR-decoded, joint stereo, single rate, 128 kbit/s
SBRdec160j	SBR-encoded, SBR-decoded, joint stereo, single rate, 160 kbit/s
SBRdec96jdl	SBR-encoded, SBR-decoded, joint stereo, dual rate, 96 kbit/s

The SBR-encoded and SBR-decoded audio items at data-rates 160 / 128 / 112 kbit/s were rated in the range “good – excellent”.

If the same SBR-encoded audio items are decoded by a Normal Layer II decoder, a significant coding degradation (penalty) is to be paid. This penalty increases if bitrate decreases. At bitrates 128 and 112 kbit/s the penalty becomes quite important – one category, i.e. about 20 points. At 96 kbit/s the penalty amounts to about two categories (“good” becomes “poor” to “bad”).

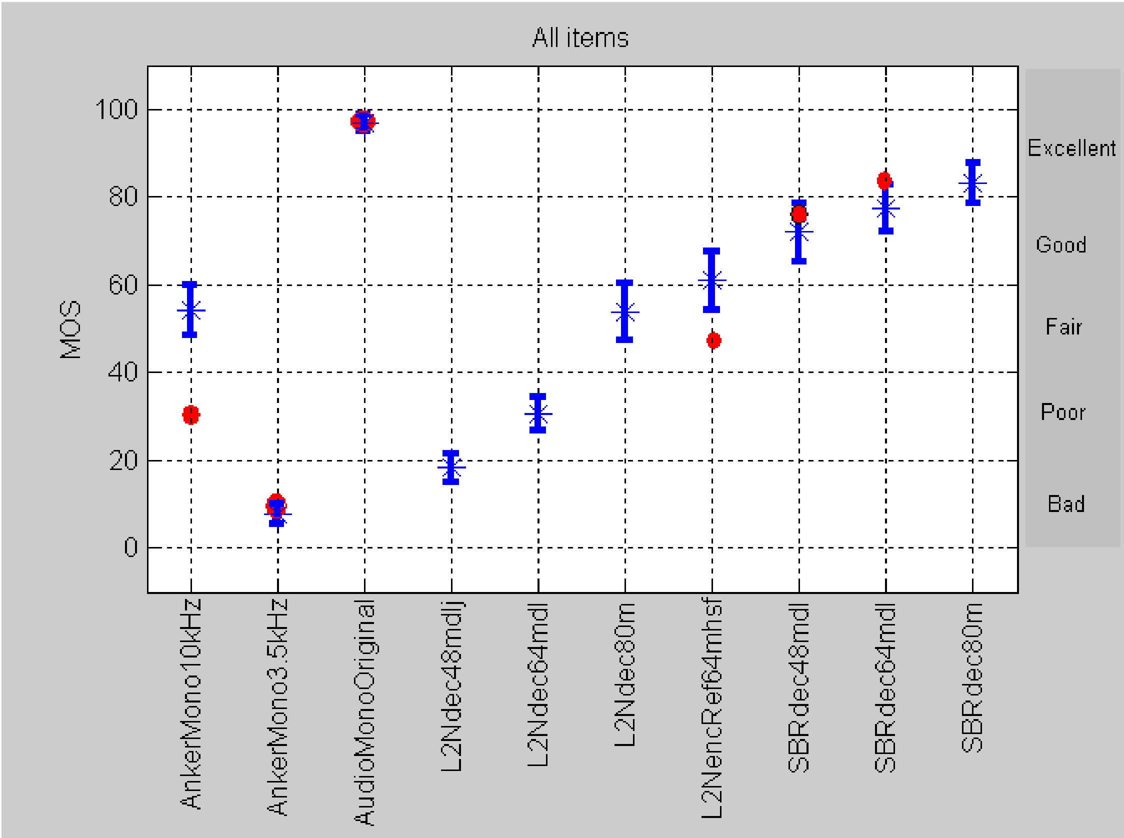
The results were compared with those found in B/AIM112 (red dots) and were found to be in good agreement.

It is interesting to note that CASE 3 (conventional Layer II case) lies almost exactly midway between CASE 1 (the highest coding gain) and CASE 2 (the lowest coding gain).

10.2 Mono test results

With 10 test subjects, the following results were found:

As in the Stereo case, CASE 2 is the worst of all three cases. This means that there is some significant coding degradation (penalty) to be paid if a normal Layer II decoder decodes SBR-encoded audio items. At a data rate of 64 kbit/s the penalty is more than two categories lower (“good” decreases to “poor”). As before, CASE 3 is almost exactly midway between CASE 1 and CASE 2.



Legend:

AnkerMono10kHz	Mono anchor 10 kHz
AnkerMono3.5kHz	Mono anchor 3.5 kHz
AudioMonoOriginal	Reference (hidden)
L2Ndec48mdl	SBR-encoded, L2-decoded, mono, dual rate, 48 kbit/s
L2Ndec64mdl	SBR-encoded, L2-decoded, mono, dual rate, 64 kbit/s
L2Ndec80m	SBR-encoded, L2-decoded, mono, single rate, 80 kbit/s
L2NencRef64mhsf	L2-encoded, L2-decoded, mono, half-sampling rate, 64 kbit/s
SBRdec48mdl	SBR-encoded, SBR-decoded, mono, dual rate, 48 kbit/s
SBRdec64mdl	SBR-encoded, SBR-decoded, mono, dual rate, 64 kbit/s
SBRdec80m	SBR-encoded, SBR-decoded, mono, single rate, 80 kbit/s

11 Summary of Part 2

11.1 Coding Gain within realistic DAB broadcast conditions

The table below shows the coding gain for different bitrates and modes under realistic broadcast conditions. The range of bitrates considered in this table is rather limited: 96 to 160 kbit/s for stereo and 80 to 96 kbit/s for mono[2]. As the bitrates in DAB are standardised and not all bitrates can be used in practice, the real coding gain is slightly less than the theoretical one.

Bit Rate MPEG Layer II	Bit Rate Layer II^{SBR} at equivalent or better audio quality	Coding Gain	Real Gain
96 kbit/s Joint Stereo	64 kbit/s Joint Stereo	40%	33%
112 kbit/s Joint Stereo	96 kbit/s Joint Stereo	33%	14%
128 kbit/s Joint Stereo	112 kbit/s Joint Stereo/Stereo	17%	13%
160 kbit/s Joint Stereo	160 kbit/s Stereo	14 %	“0%”
80 kbit/s Mono	48 kbit/s Mono	50%	40%
96 kbit/s Mono	80 kbit/s Mono	22%	17%

The coding gain for 64 kbit/s could not be measured directly because the Coding Technology software did not allow for this bitrate. However, it was possible to assess the coding gain at 64 kbit/s by interpolation as follows:

64 kbit/s Mono	40 kbit/s Mono	60%	50%

11.2 Loss of Quality in existing DAB Receivers

Existing DAB services using full sample rate (that is, not using LSF - lower sampling rate) will loose the high frequencies if the DAB services start to employ SBR. At lower bit rates the quality of audio produced by normal Layer 2 decoders will be degraded because proportionally fewer bits will be allocated to the backwards compatible part of the bit stream. This penalty increases if bit rate decreases.

If broadcasts are already being made using LSF mode, then the quality degradation suffered by the legacy receiver following the introduction of SBR will be negligible. However, the receiver would have to be upgraded to have a SBR compatible decoder for the listener to hear the benefit of SBR.

[2] The coding gain for 192 kbit/s stereo and 64 kbit/s mono were not measured directly.

12 References

[1] Christian Kain, „Untersuchungen zur Verbesserung der Audiocodierung bei DAB mit dem kompatiblen Verfahren der Spectral Band Replication“, Diplomarbeit IRT 2002 (diploma thesis) EBU Document B/AIM112 (in German)

[2] Report of the 14th meeting of EBU Project Group B/AIM, document B/AIM128

[3] Gilchrist et al. : EUREKA147 : Test of Error Performance of the DAB System BBC, BBC RD 1994/18

[4] Lever, Richard, Gilchrist: EUREKA147 : Subjective Assessment of the Error Performance of the DAB System, including tests at 24 kHz Audio Sampling Frequency, BBC RD 1996/7

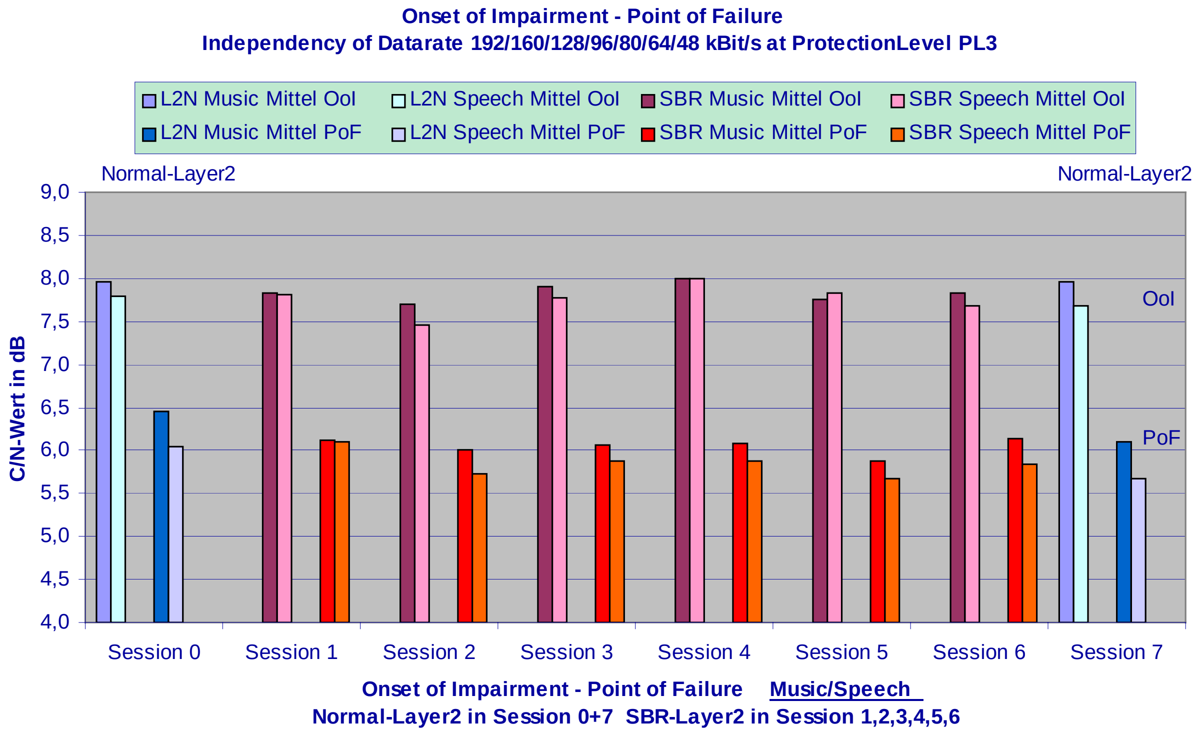
[5] Wolfgang Krafft (IRT): Subjective assessment of error performance of the DAB audio coding system according to the compatible method of Spectral Band Replication (SBR) in comparison with the normal MPEG Layer 2 coding, EBU Project group B/AIM 143, July 2004

[6] B/CASE 100 BMC 477: „Subjective Audio Quality Achievable at Various Bit-rates for MPEG-Audio Layer II and Layer III” (Contribution from Project Group B/CASE, March 1999)

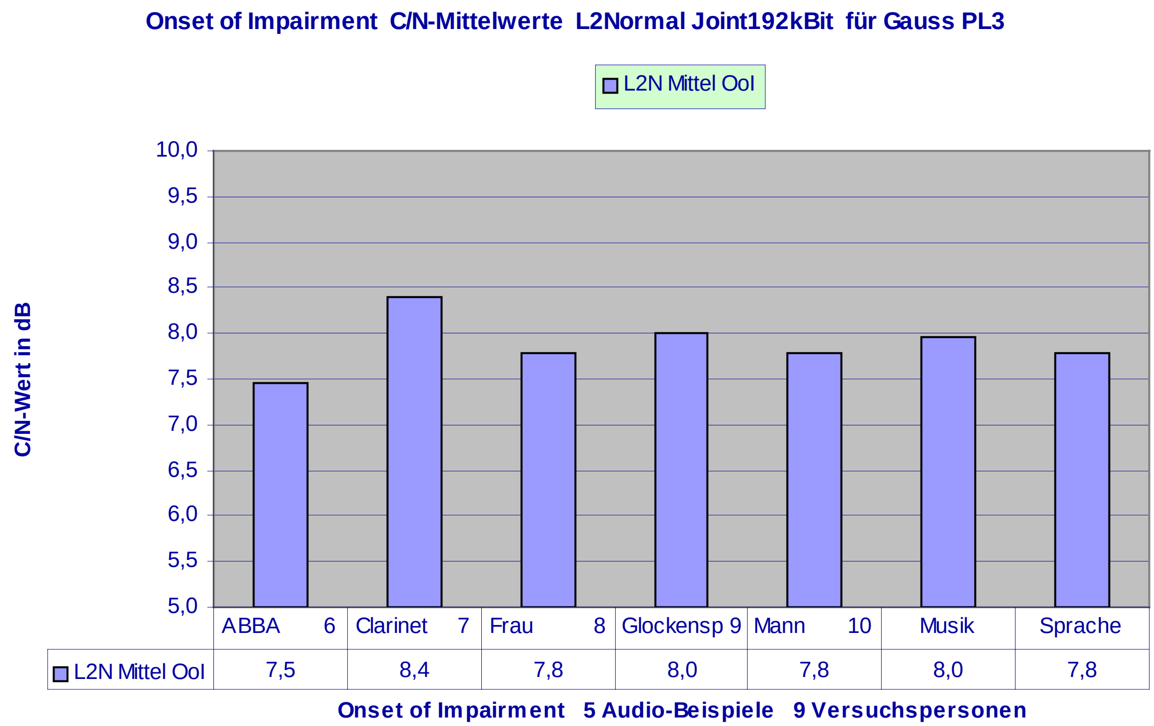
13 Appendix 1: Results on Onset-of-Impairment and Point-of-Failure

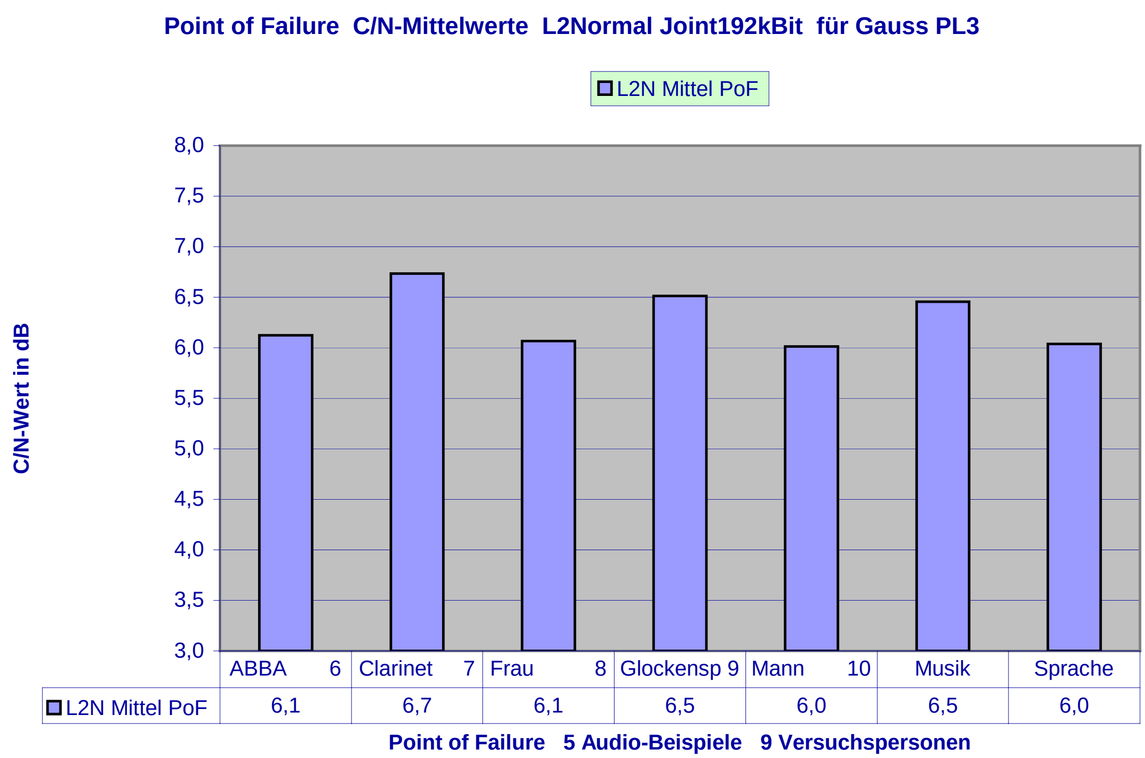
13.1 Dependency on Data Rate in Gaussian-Channel at Protection Level PL = 3

The following picture shows the result in a global consideration, depicted from single diagrams.

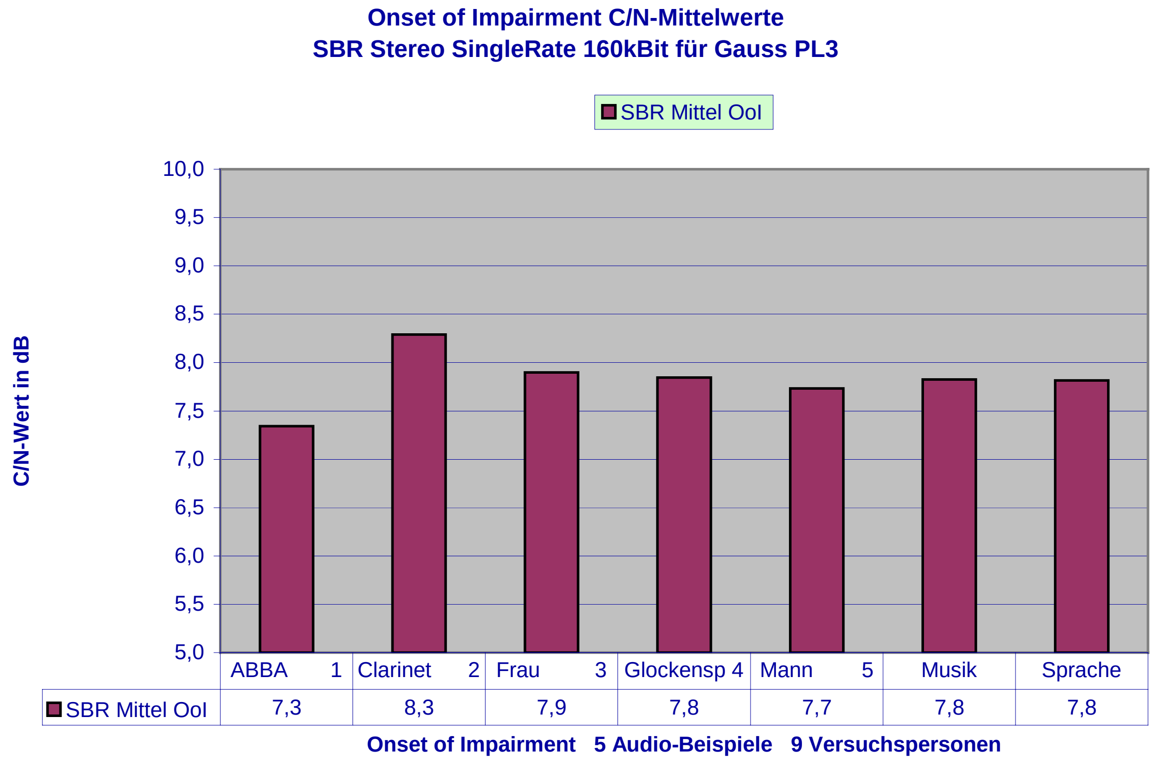


Session 0

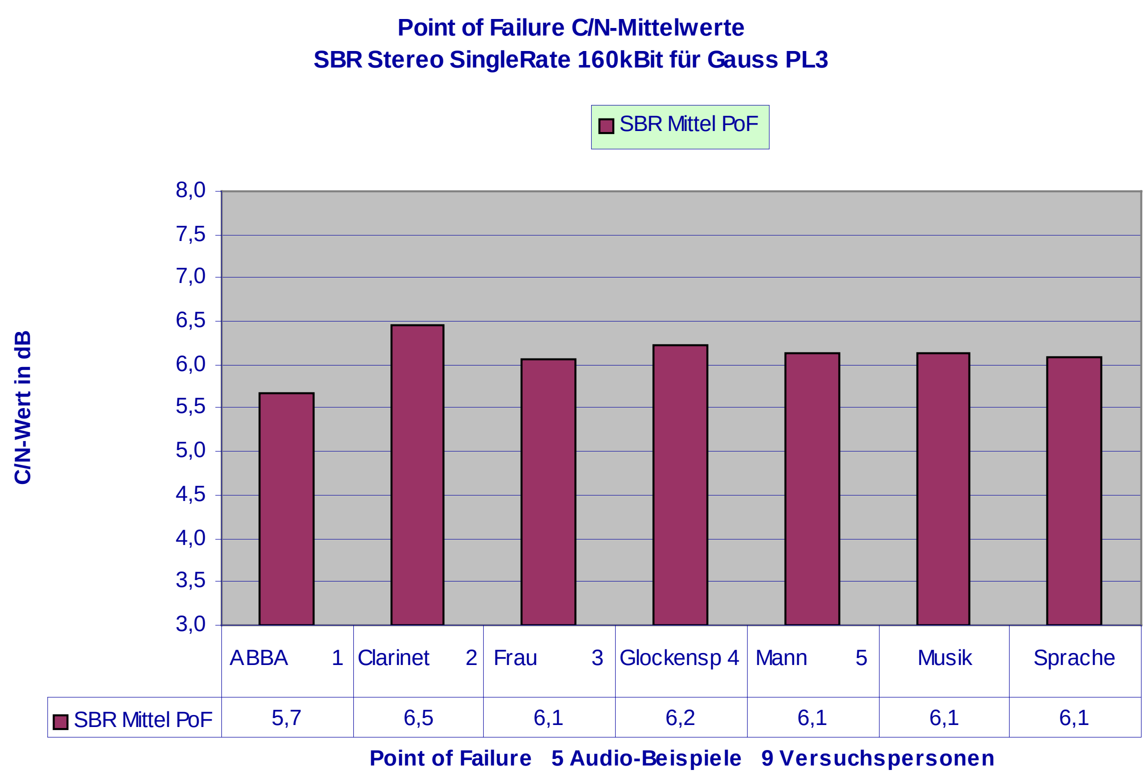

Onset of Impairment C/N-Mittelwerte L2Normal Joint192kBit für Gauss PL3
L2N Mittel OoI
C/N-Wert in dB
10,0
9,5
9,0
8,5
8,0
7,5
7,0
6,5
6,0
5,5
5,0
ABBA 6
Clarinet 7
Frau 8
Glockensp 9
Mann 10
Musik
Sprache
L2N Mittel OoI
7,5
8,4
7,8
8,0
7,8
8,0
7,8
Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen


Point of Failure C/N-Mittelwerte L2Normal Joint192kBit für Gauss PL3
L2N Mittel PoF
C/N-Wert in dB
8,0
7,5
7,0
6,5
6,0
5,5
5,0
4,5
4,0
3,5
3,0
ABBA 6
Clarinet 7
Frau 8
Glockensp 9
Mann 10
Musik
Sprache
L2N Mittel PoF
6,1
6,7
6,1
6,5
6,0
6,5
6,0
Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Session 1



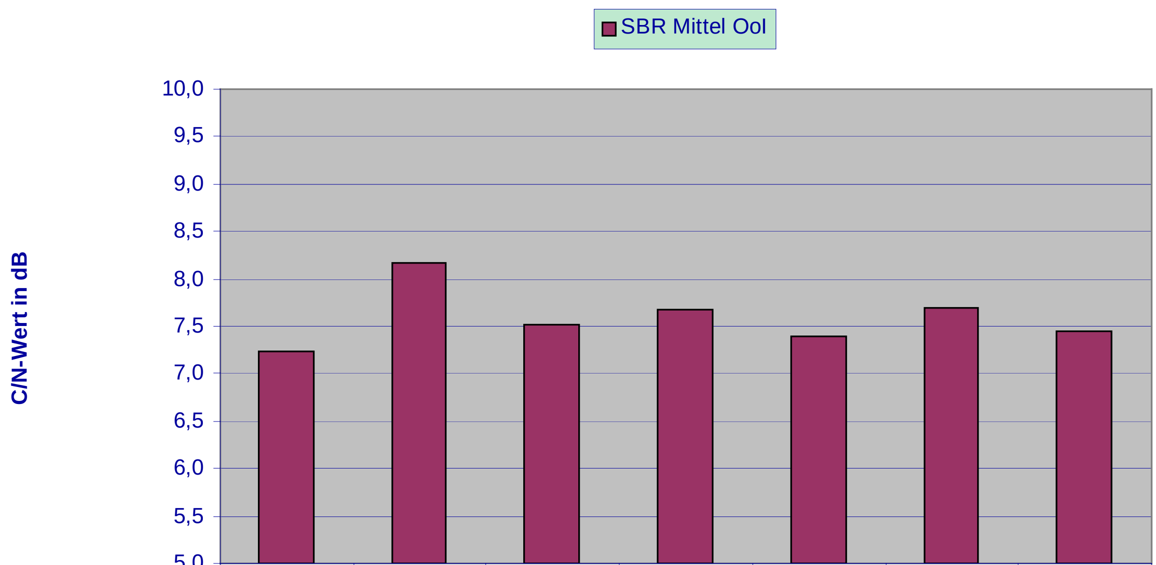
	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel OoI	7,3	8,3	7,9	7,8	7,7	7,8	7,8



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel PoF	5,7	6,5	6,1	6,2	6,1	6,1	6,1

Session 2

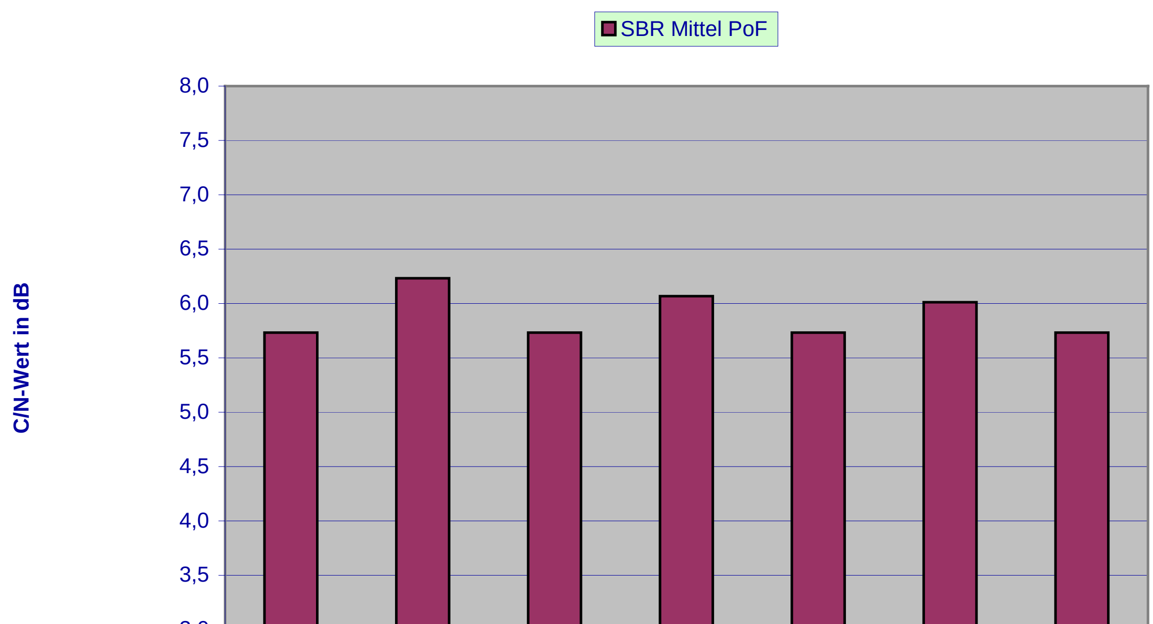
Onset of Impairment C/N-Mittelwerte SBR Joint DualRate 128kBit für Gauss PL3



	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
SBR Mittel OoI	7,2	8,2	7,5	7,7	7,4	7,7	7,5

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

Point of Failure C/N-Mittelwerte SBR Joint DualRate 128kBit für Gauss PL3

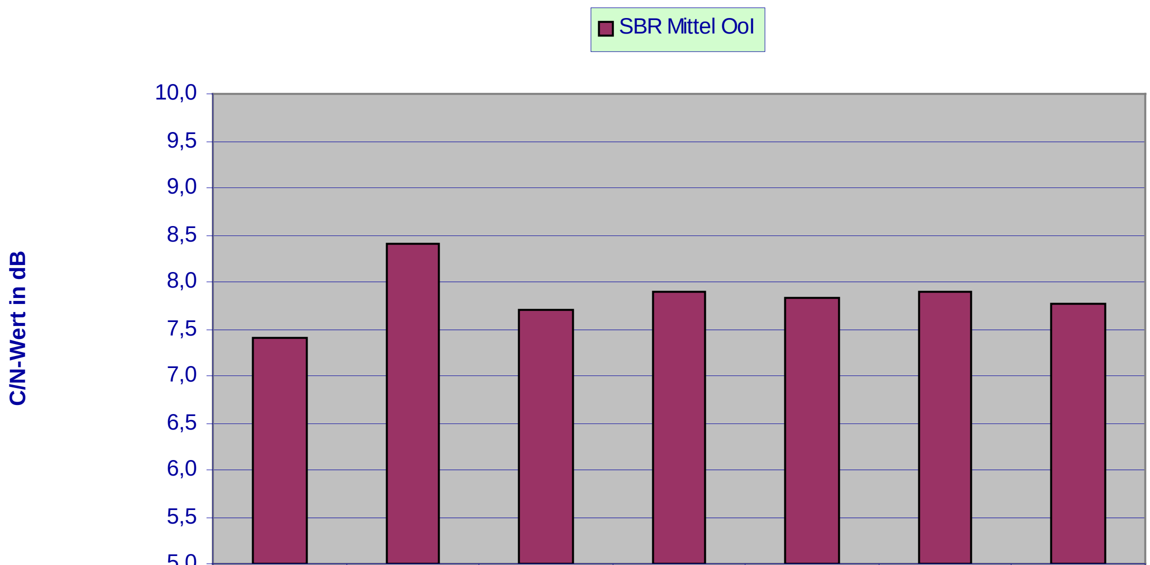


	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
SBR Mittel PoF	5,7	6,2	5,7	6,1	5,7	6,0	5,7

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Session 3

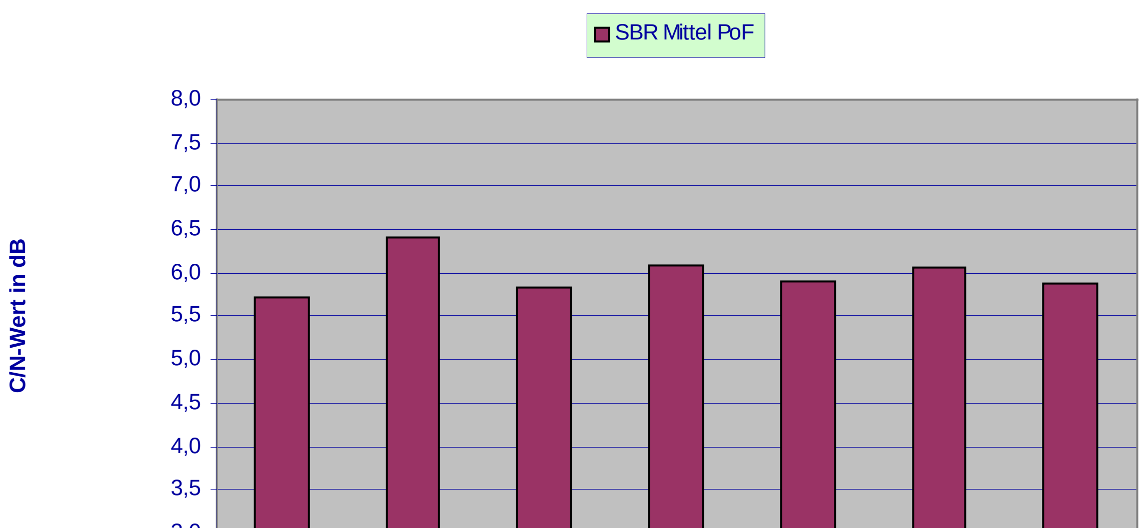
Onset of Impairment C/N-Mittelwerte
SBR Joint DualRate LowComp 96kBit für Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel OoI	7,4	8,4	7,7	7,9	7,8	7,9	7,8

Onset of Impairment 5 Audio-Beispiele 8 Versuchspersonen

Point of Failure C/N-Mittelwerte
SBR Joint DualRate LowComp 96kBit für Gauss PL3

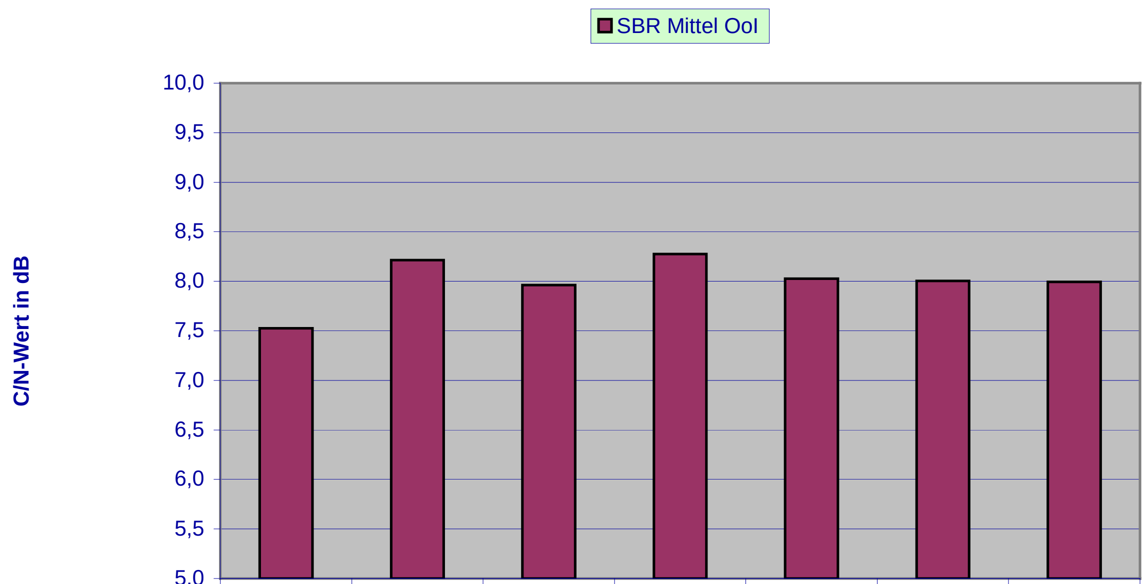


	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel PoF	5,7	6,4	5,8	6,1	5,9	6,1	5,9

Point of Failure 5 Audio-Beispiele 8 Versuchspersonen

Session 4

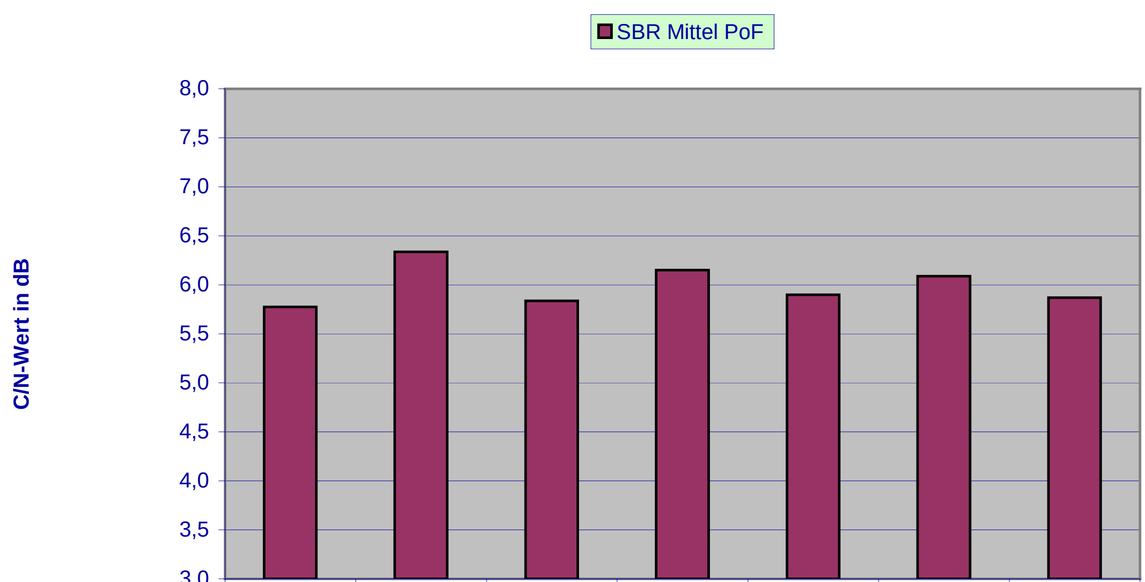
Onset of Impairment C/N-Mittelwerte
SBR Mono SingleRate 80kBit für Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel OoI	7,5	8,2	8,0	8,3	8,0	8,0	8,0

Onset of Impairment 5 Audio-Beispiele 8 Versuchspersonen

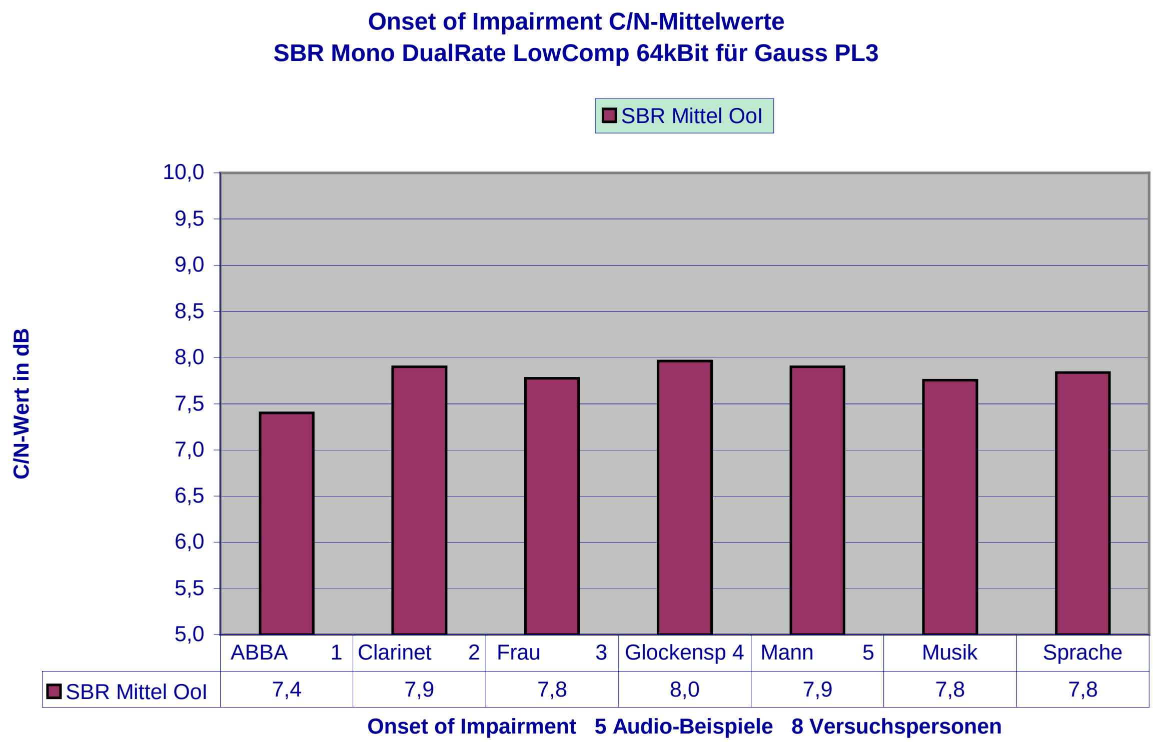
Point of Failure C/N-Mittelwerte
SBR Mono SingleRate 80kBit für Gauss PL3

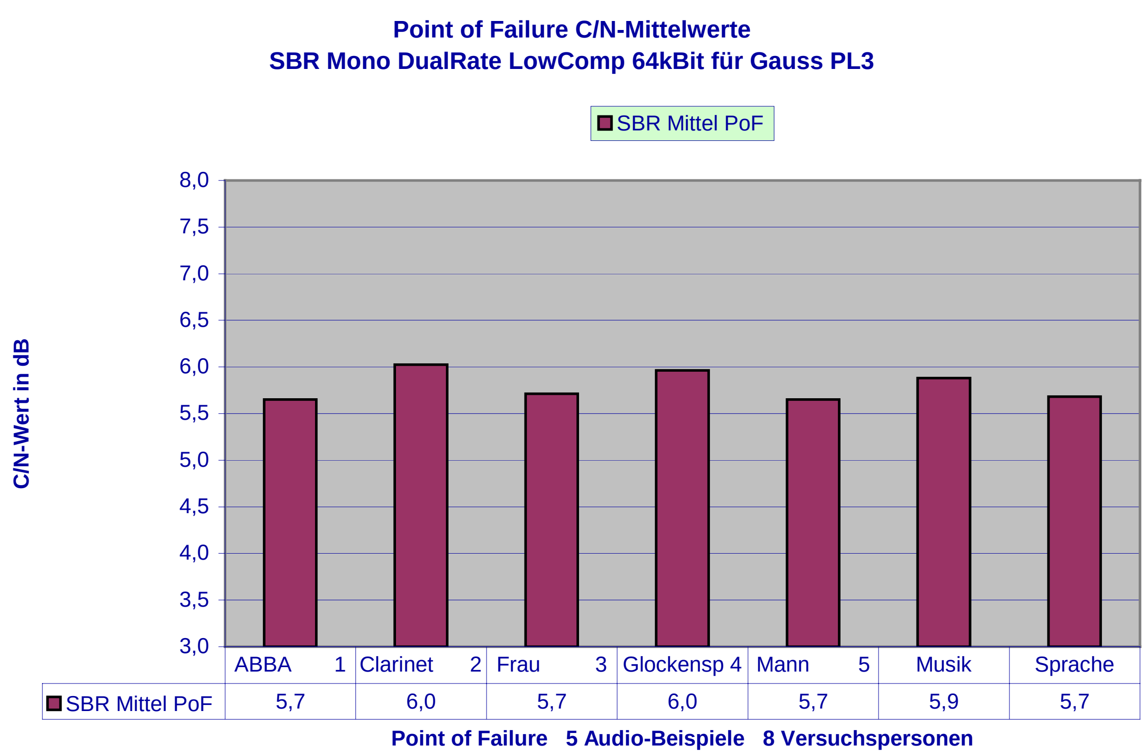


	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel PoF	5,8	6,3	5,8	6,2	5,9	6,1	5,9

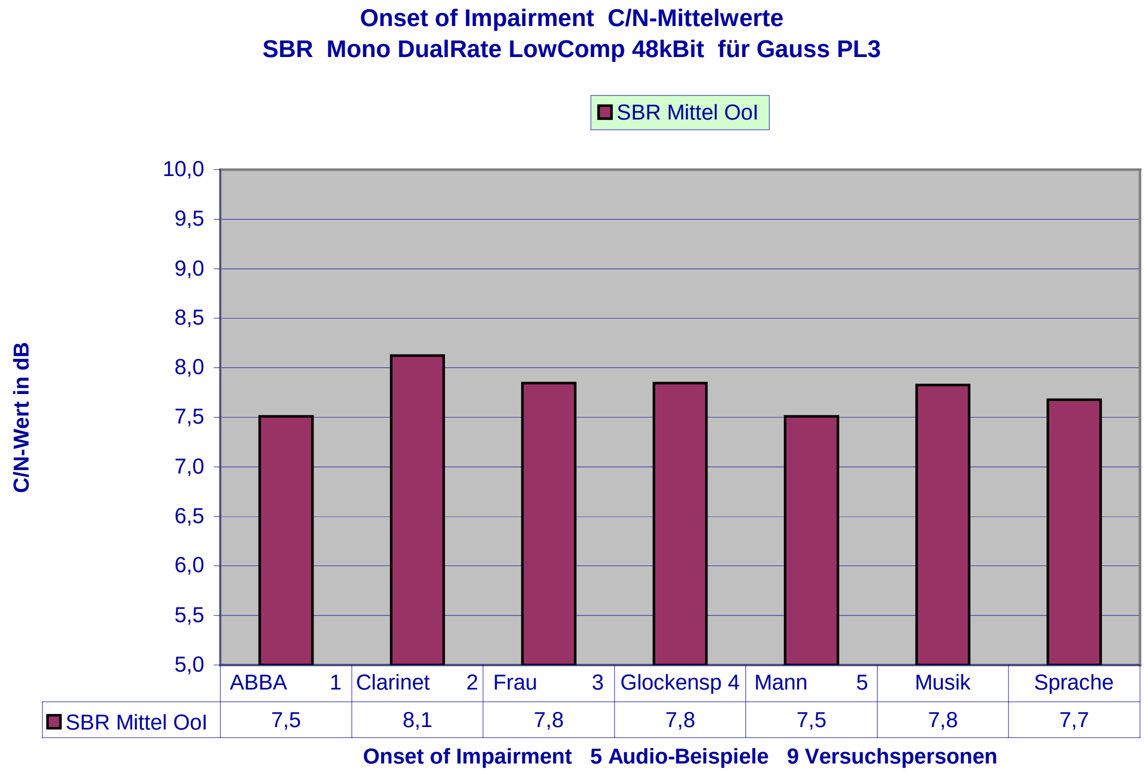
Point of Failure 5 Audio-Beispiele 8 Versuchspersonen

Session 5

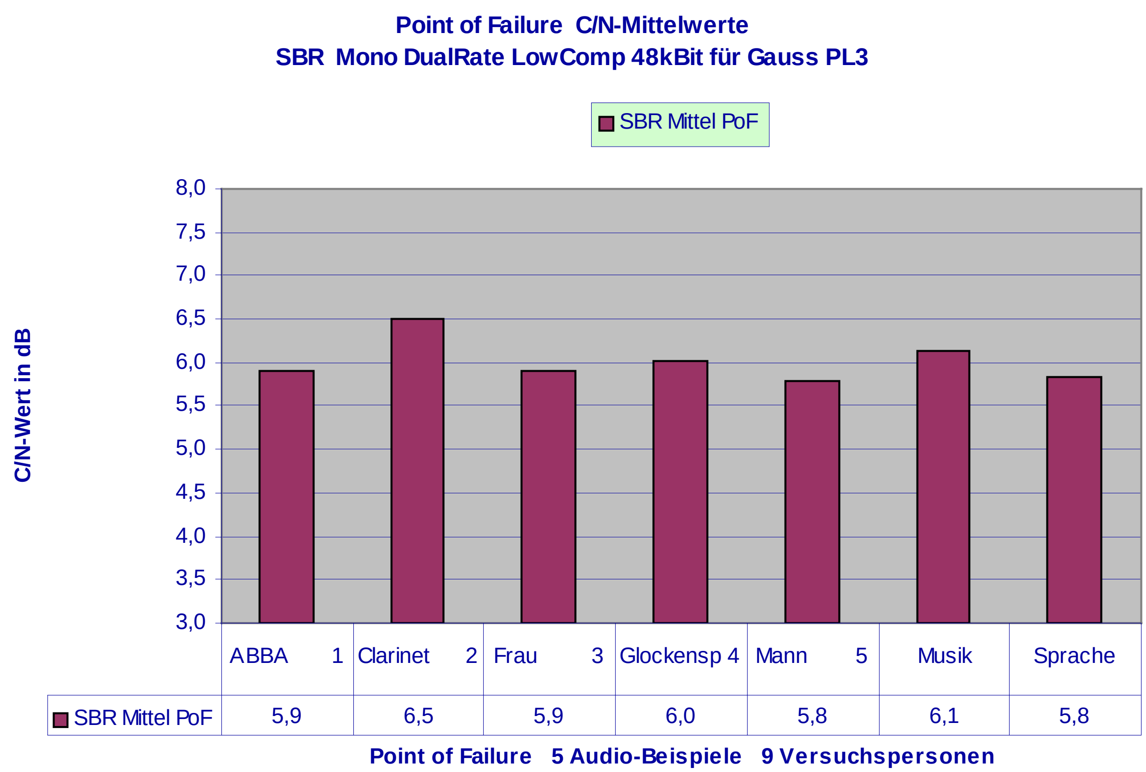

Onset of Impairment C/N-Mittelwerte
SBR Mono DualRate LowComp 64kBit für Gauss PL3
SBR Mittel OoI
C/N-Wert in dB
10,0
9,5
9,0
8,5
8,0
7,5
7,0
6,5
6,0
5,5
5,0
ABBA 1
Clarinet 2
Frau 3
Glockensp 4
Mann 5
Musik
Sprache
SBR Mittel OoI
7,4
7,9
7,8
8,0
7,9
7,8
7,8
Onset of Impairment 5 Audio-Beispiele 8 Versuchspersonen


Point of Failure C/N-Mittelwerte
SBR Mono DualRate LowComp 64kBit für Gauss PL3
SBR Mittel PoF
C/N-Wert in dB
8,0
7,5
7,0
6,5
6,0
5,5
5,0
4,5
4,0
3,5
3,0
ABBA 1
Clarinet 2
Frau 3
Glockensp 4
Mann 5
Musik
Sprache
SBR Mittel PoF
5,7
6,0
5,7
6,0
5,7
5,9
5,7
Point of Failure 5 Audio-Beispiele 8 Versuchspersonen

Session 6



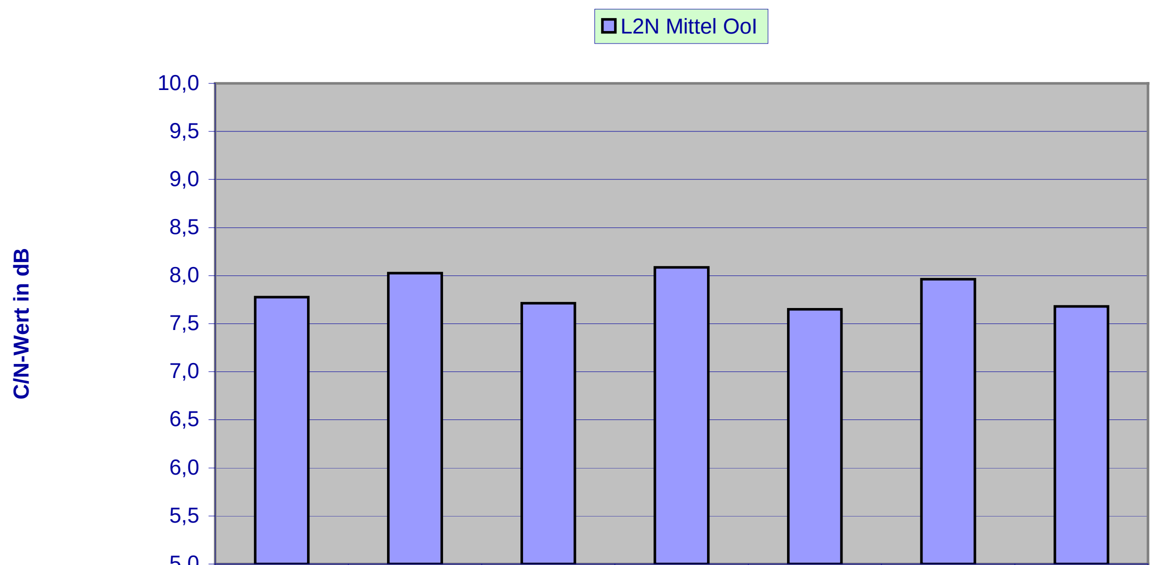
	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel OoI	7,5	8,1	7,8	7,8	7,5	7,8	7,7



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
SBR Mittel PoF	5,9	6,5	5,9	6,0	5,8	6,1	5,8

Session 7

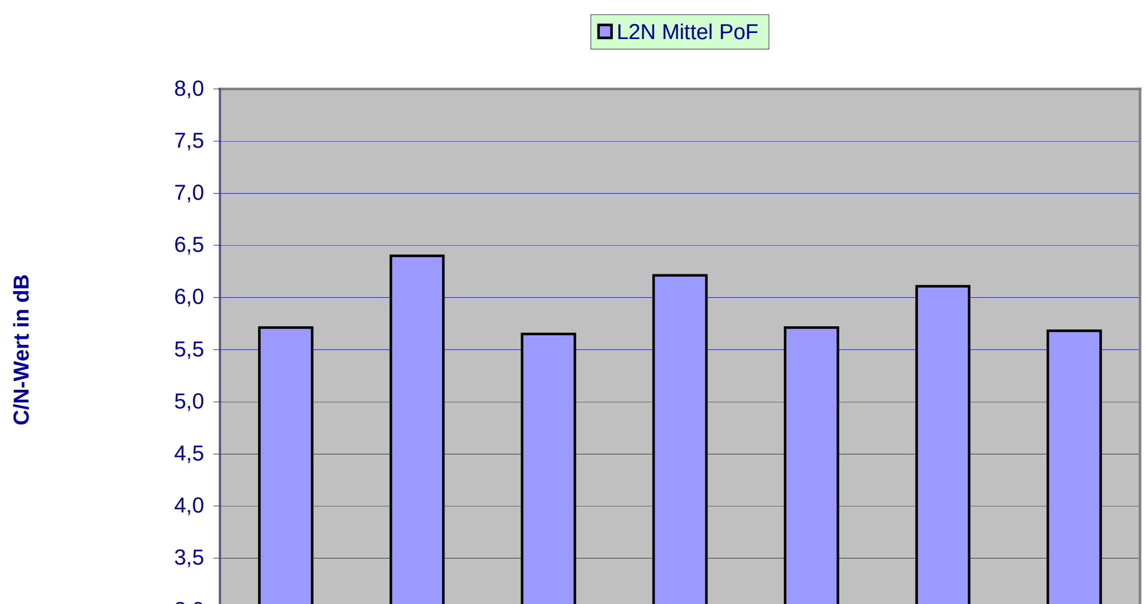
Onset of Impairment C/N-Mittelwerte L2Normal Mono48kBit für Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel OoI	7,8	8,0	7,7	8,1	7,7	8,0	7,7

Onset of Impairment 5 Audio-Beispiele 8 Versuchspersonen

Point of Failure C/N-Mittelwerte L2Normal Mono48kBit für Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel PoF	5,7	6,4	5,7	6,2	5,7	6,1	5,7

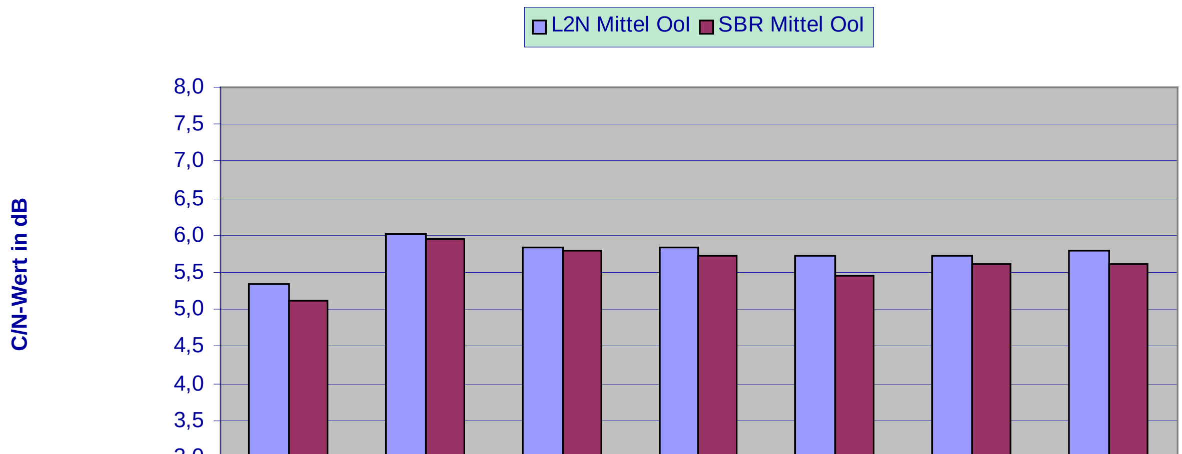
Point of Failure 5 Audio-Beispiele 8 Versuchspersonen

13.2 Dependency on Protection Level PL = 1 / 3 / 5 in Gaussian-Channel

Session 0 + Session 2

Normal-Layer2 (192 kBit/s Session 0) and SBR-Layer2 (128 kBit/s Session 2)

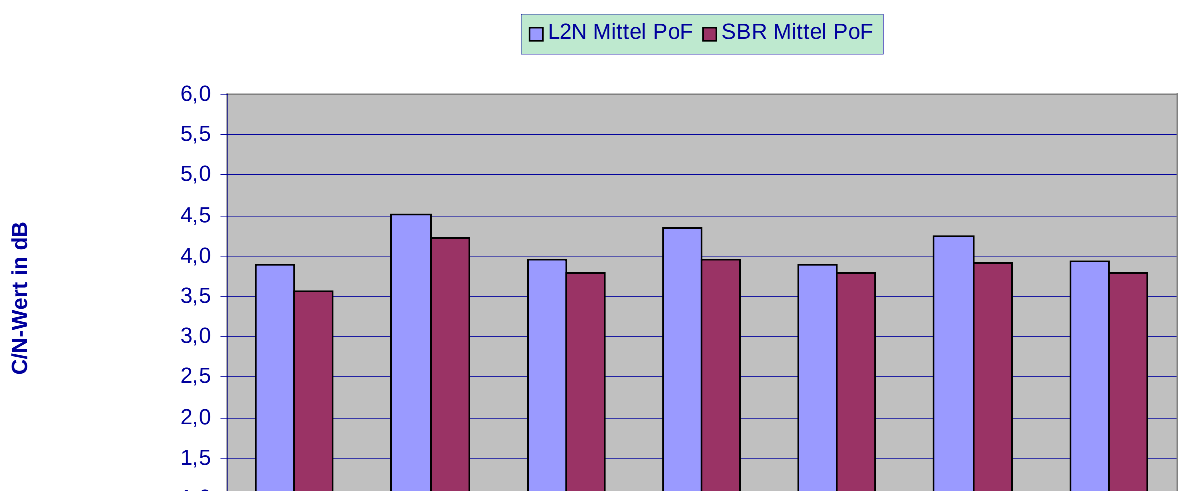
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL1



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel OoI	5,3	6,0	5,8	5,8	5,7	5,7	5,8
SBR Mittel OoI	5,1	6,0	5,8	5,7	5,5	5,6	5,6

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

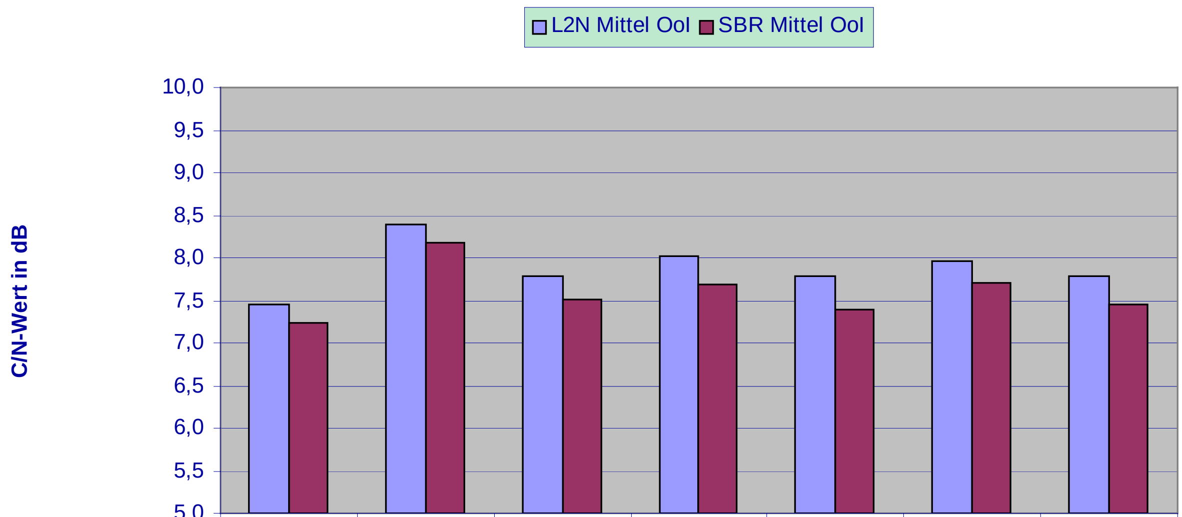
Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL1



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel PoF	3,9	4,5	4,0	4,3	3,9	4,3	3,9
SBR Mittel PoF	3,6	4,2	3,8	4,0	3,8	3,9	3,8

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

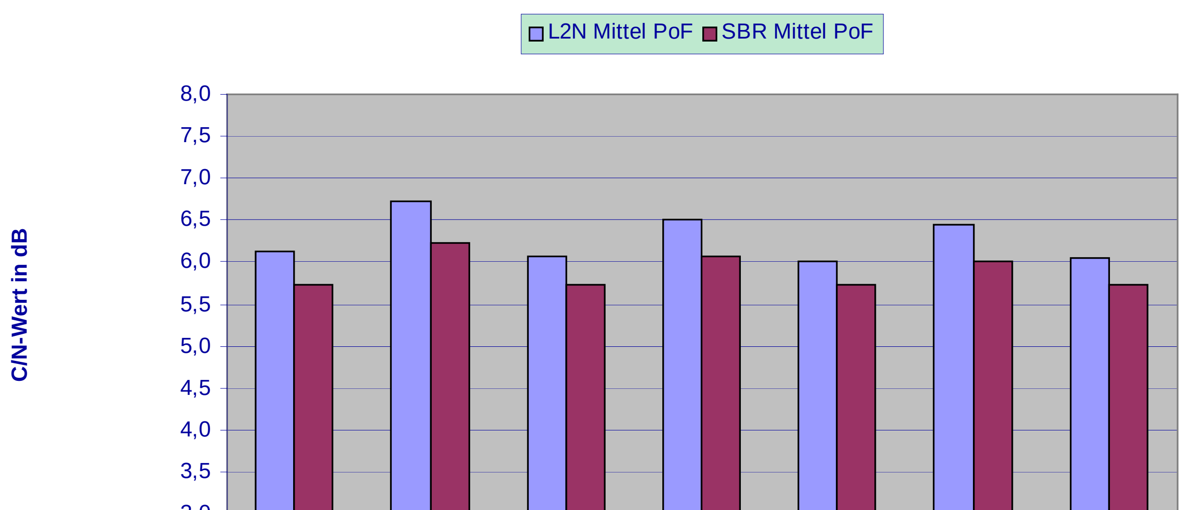
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL3



	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
L2N Mittel OoI	7,5	8,4	7,8	8,0	7,8	8,0	7,8
SBR Mittel OoI	7,2	8,2	7,5	7,7	7,4	7,7	7,5

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

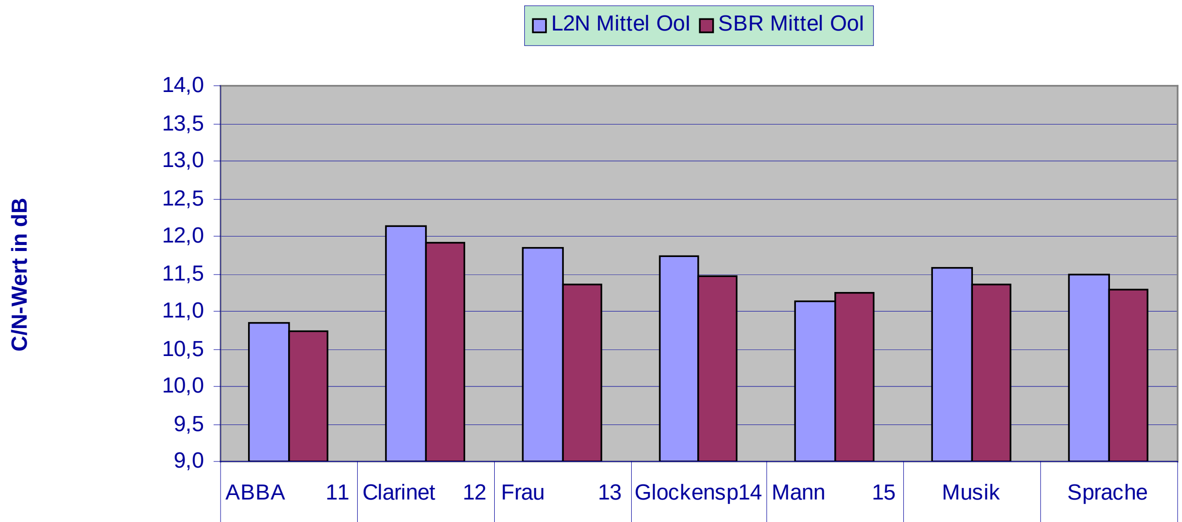
Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL3



	ABBA 6	Clarinet 7	Frau 8	Glockensp 9	Mann 10	Musik	Sprache
L2N Mittel PoF	6,1	6,7	6,1	6,5	6,0	6,5	6,0
SBR Mittel PoF	5,7	6,2	5,7	6,1	5,7	6,0	5,7

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

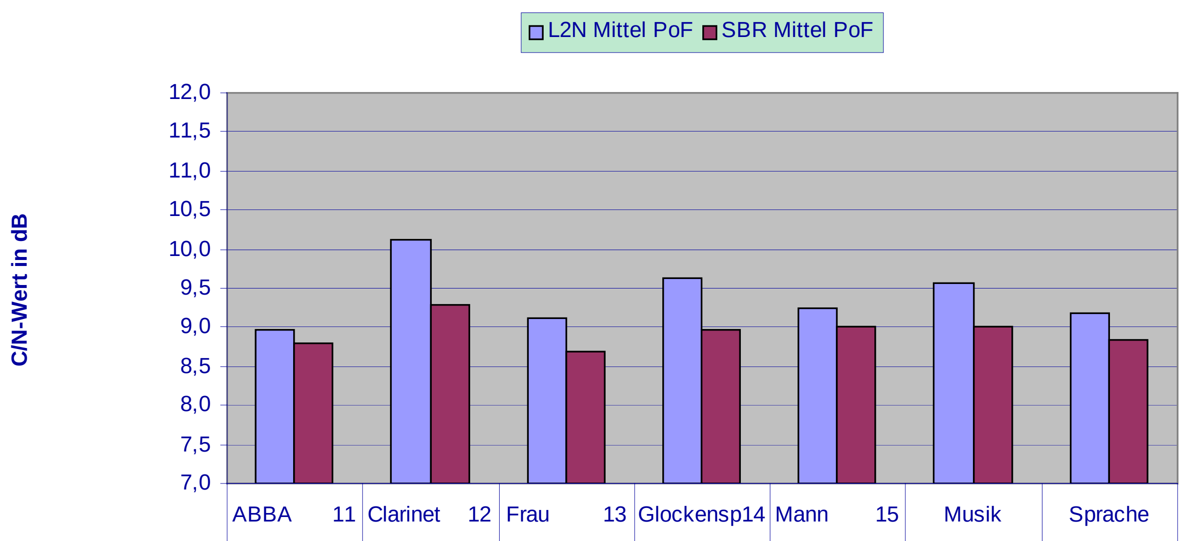
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL5



	ABBA 11	Clarinet 12	Frau 13	Glockensp14	Mann 15	Musik	Sprache
L2N Mittel OoI	10,8	12,1	11,8	11,7	11,1	11,6	11,5
SBR Mittel OoI	10,7	11,9	11,3	11,5	11,2	11,4	11,3

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Gauss PL5



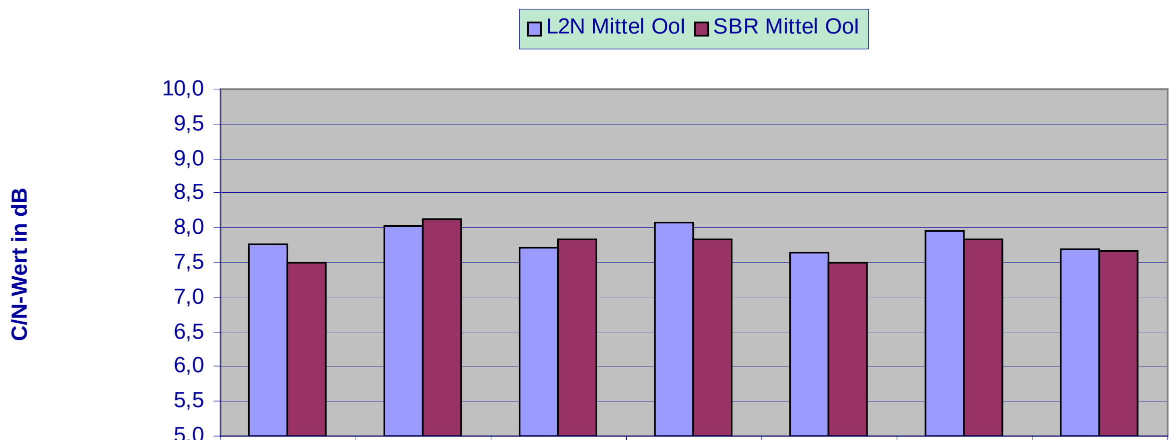
	ABBA 11	Clarinet 12	Frau 13	Glockensp14	Mann 15	Musik	Sprache
L2N Mittel PoF	9,0	10,1	9,1	9,6	9,2	9,6	9,2
SBR Mittel PoF	8,8	9,3	8,7	9,0	9,0	9,0	8,8

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Session 7 + Session 6

Normal-Layer2 (48 kBit/s Session 7) and SBR-Layer2 (48 kBit/s Session 6) only ProtectionLevel PL = 3

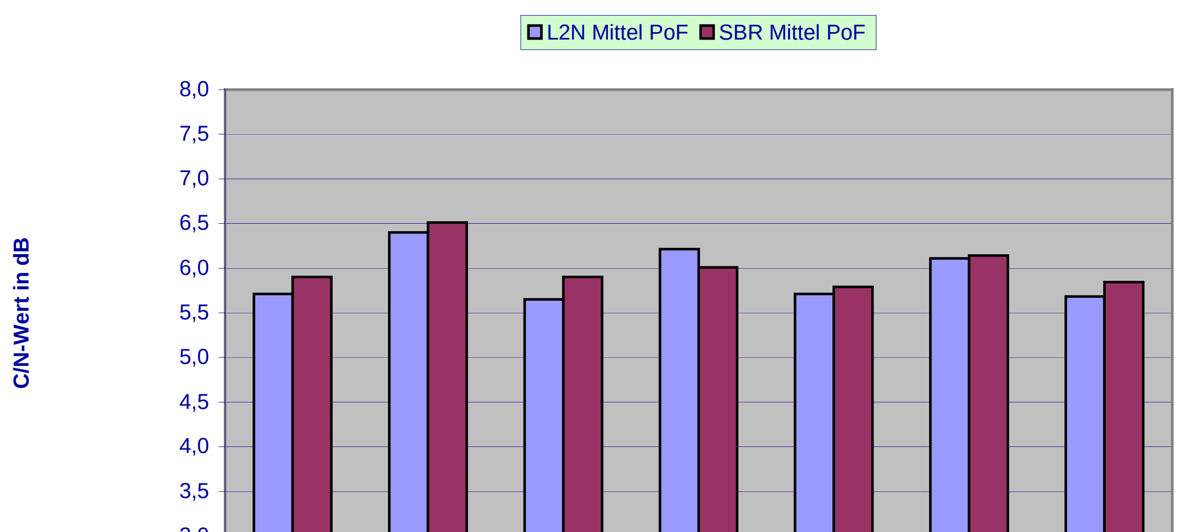
Onset of Impairment C/N-Mittelwerte L2Normal und SBR Mono48kBit Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel OoI	7,8	8,0	7,7	8,1	7,7	8,0	7,7
SBR Mittel OoI	7,5	8,1	7,8	7,8	7,5	7,8	7,7

Onset of Impairment 5 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

Point of Failure C/N-Mittelwerte L2Normal und SBR Mono48kBit Gauss PL3



	ABBA 1	Clarinet 2	Frau 3	Glockensp 4	Mann 5	Musik	Sprache
L2N Mittel PoF	5,7	6,4	5,7	6,2	5,7	6,1	5,7
SBR Mittel PoF	5,9	6,5	5,9	6,0	5,8	6,1	5,8

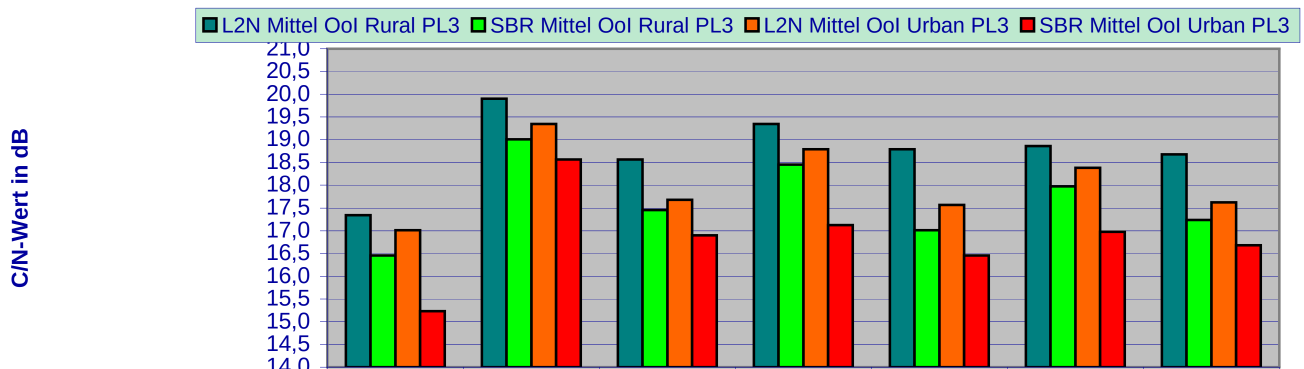
Point of Failure 5 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

13.3 Rayleigh-Channel – Rural/Urban at Protection Level PL = 3

Session 0 + Session 2

Normal-Layer2 (192 kBit/s Session 0) and SBR-Layer2 (128 kBit/s Session 2)

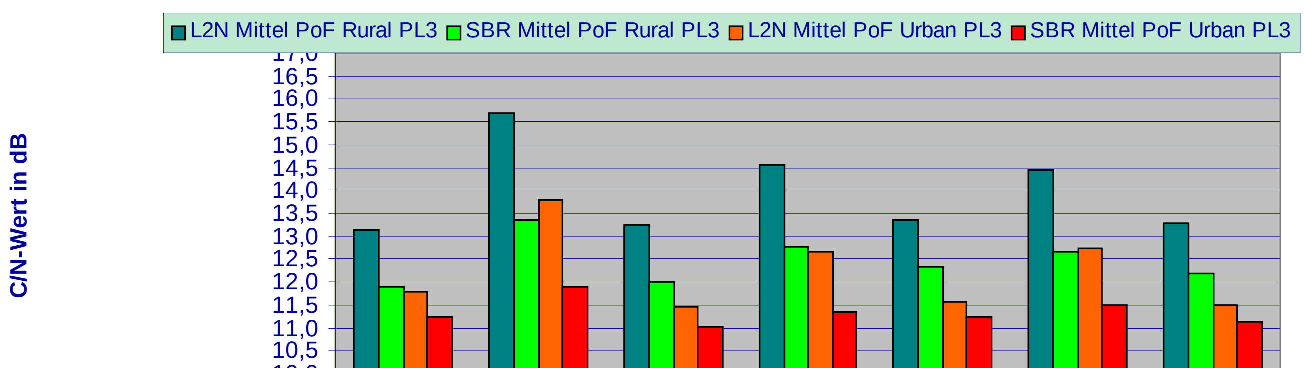
Onset of Impairment C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Rural und Urban PL3



	ABBA 21	Clarinet 22	Frau 23	Glockensp24	Mann 25	Musik	Sprache
L2N Mittel OoI Rural PL3	17,3	19,9	18,6	19,3	18,8	18,9	18,7
SBR Mittel OoI Rural PL3	16,5	19,0	17,5	18,5	17,0	18,0	17,2
L2N Mittel OoI Urban PL3	17,0	19,3	17,7	18,8	17,6	18,4	17,6
SBR Mittel OoI Urban PL3	15,2	18,6	16,9	17,1	16,5	17,0	16,7

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

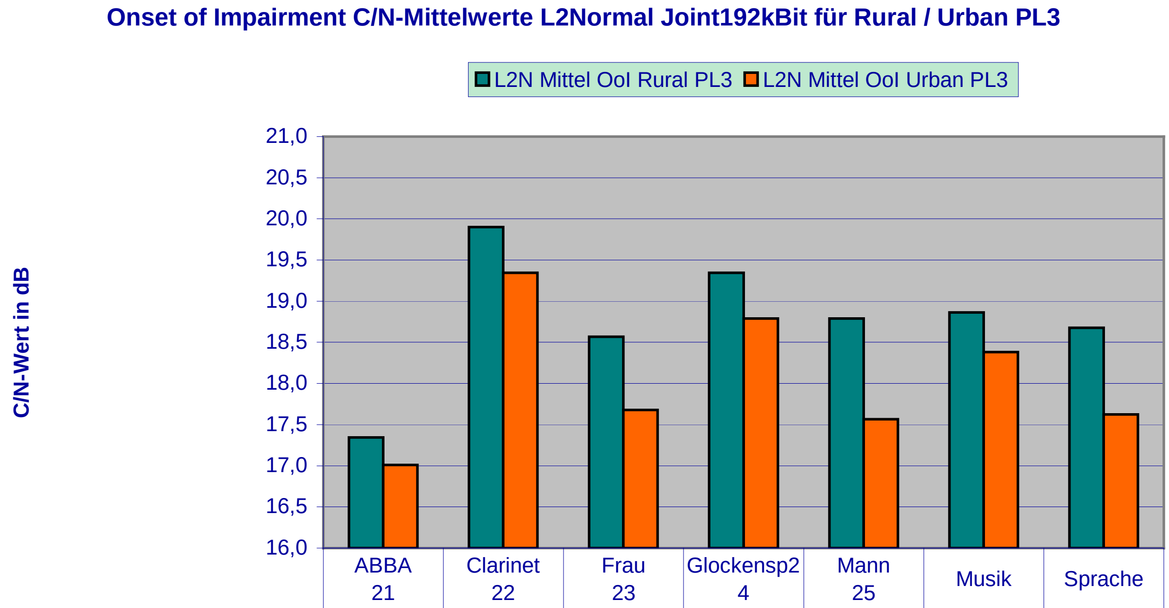
Point of Failure C/N-Mittelwerte
L2Normal Joint192kBit <---> SBR Joint DualRate 128kBit für Rural und Urban PL3



	ABBA 21	Clarinet 22	Frau 23	Glockensp24	Mann 25	Musik	Sprache
L2N Mittel PoF Rural PL3	13,1	15,7	13,2	14,6	13,3	14,5	13,3
SBR Mittel PoF Rural PL3	11,9	13,3	12,0	12,8	12,3	12,7	12,2
L2N Mittel PoF Urban PL3	11,8	13,8	11,5	12,7	11,6	12,8	11,5
SBR Mittel PoF Urban PL3	11,2	11,9	11,0	11,3	11,2	11,5	11,1

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

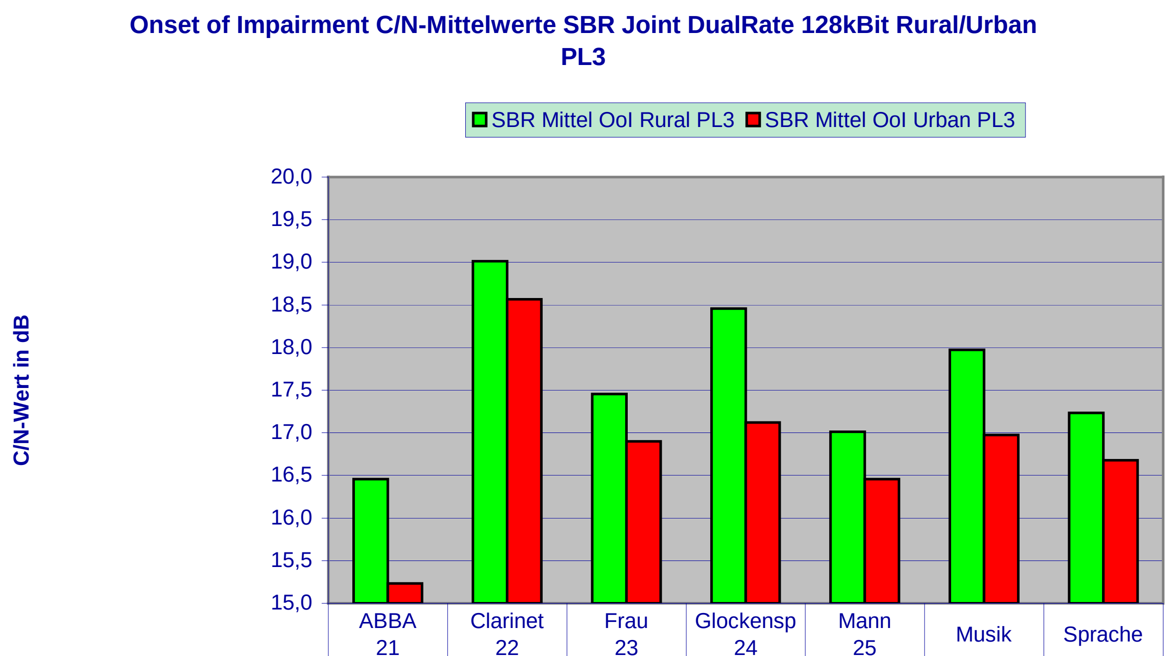
Session 0



	ABBA 21	Clarinet 22	Frau 23	Glockensp2 4	Mann 25	Musik	Sprache
L2N Mittel OoI Rural PL3	17,3	19,9	18,6	19,3	18,8	18,9	18,7
L2N Mittel OoI Urban PL3	17,0	19,3	17,7	18,8	17,6	18,4	17,6

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

Session 2

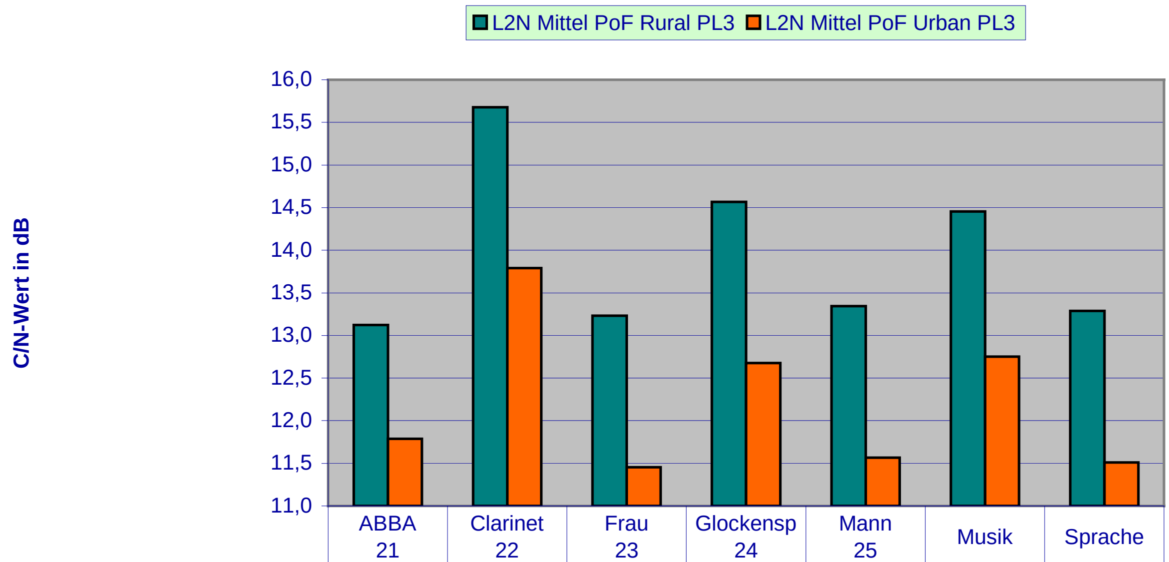


	ABBA 21	Clarinet 22	Frau 23	Glockensp 24	Mann 25	Musik	Sprache
SBR Mittel OoI Rural PL3	16,5	19,0	17,5	18,5	17,0	18,0	17,2
SBR Mittel OoI Urban PL3	15,2	18,6	16,9	17,1	16,5	17,0	16,7

Onset of Impairment 5 Audio-Beispiele 9 Versuchspersonen

Session 0

Point of Failure C/N-Mittelwerte L2Normal Joint192kBit für Rural / Urban PL3

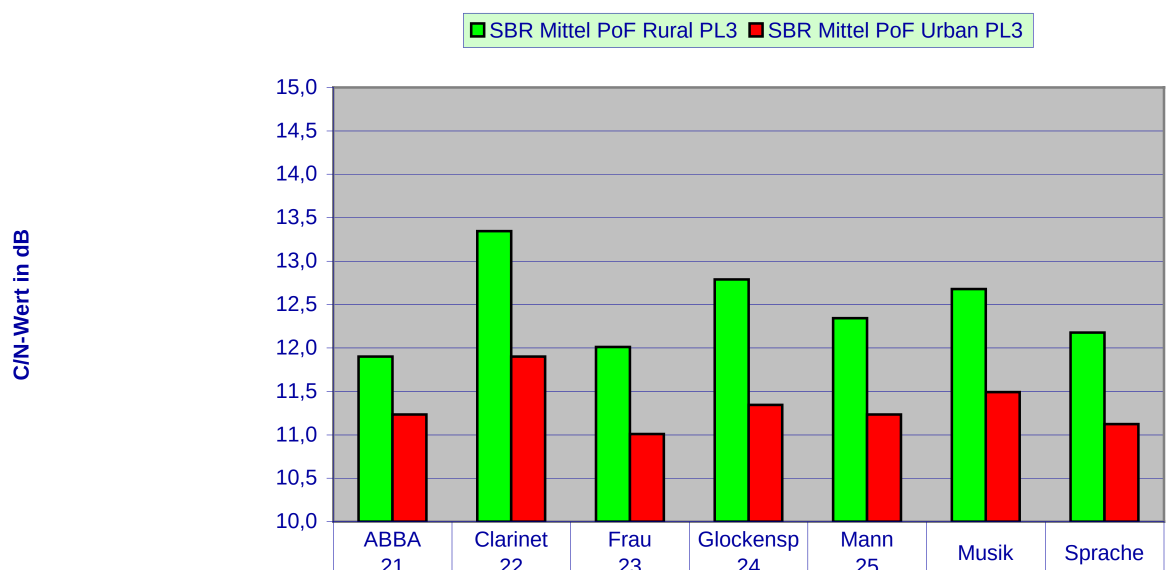


	ABBA 21	Clarinet 22	Frau 23	Glockensp 24	Mann 25	Musik	Sprache
L2N Mittel PoF Rural PL3	13,1	15,7	13,2	14,6	13,3	14,5	13,3
L2N Mittel PoF Urban PL3	11,8	13,8	11,5	12,7	11,6	12,8	11,5

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Session 2

Point of Failure C/N-Mittelwerte SBR Joint DualRate 128kBit für Rural / Urban PL3



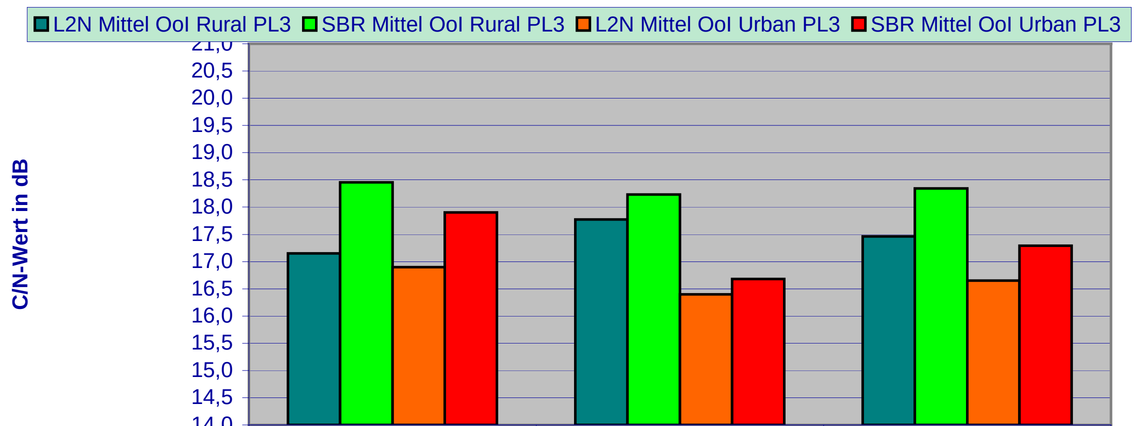
	ABBA 21	Clarinet 22	Frau 23	Glockensp 24	Mann 25	Musik	Sprache
SBR Mittel PoF Rural PL3	11,9	13,3	12,0	12,8	12,3	12,7	12,2
SBR Mittel PoF Urban PL3	11,2	11,9	11,0	11,3	11,2	11,5	11,1

Point of Failure 5 Audio-Beispiele 9 Versuchspersonen

Session 7 + Session 6

Normal-Layer2 (48 kBit/s Session 7) and SBR-Layer2 (48 kBit/s Session 6)

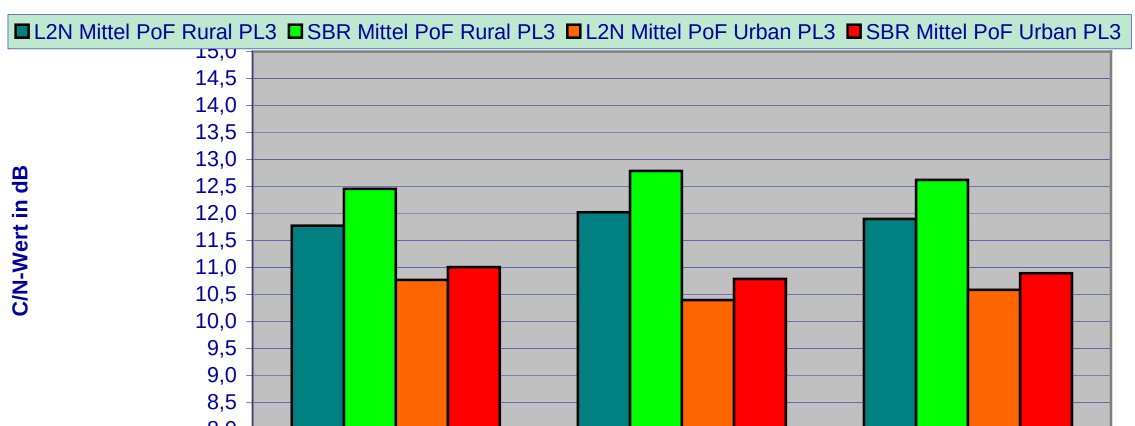
Onset of Impairment C/N-Mittelwerte L2Normal und SBR Mono48kBit Rural und Urban PL3



	Frau 8	Mann 9	Sprache
L2N Mittel OoI Rural PL3	17,2	17,8	17,5
SBR Mittel OoI Rural PL3	18,5	18,2	18,3
L2N Mittel OoI Urban PL3	16,9	16,4	16,7
SBR Mittel OoI Urban PL3	17,9	16,7	17,3

Onset of Impairment 2 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

Point of Failure C/N-Mittelwerte L2Normal und SBR Mono48kBit Rural und Urban PL3

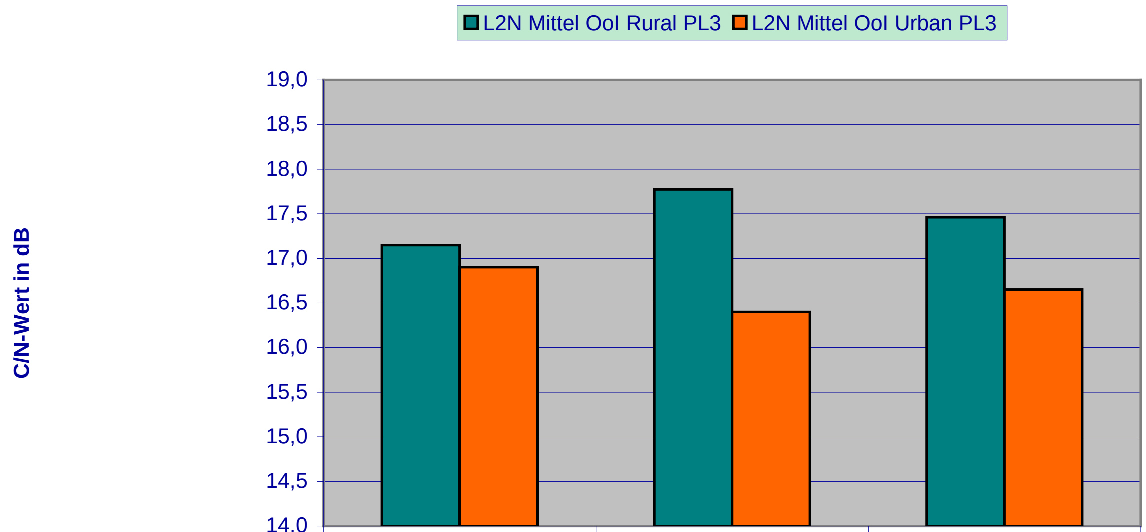


	Frau 8	Mann 9	Sprache
L2N Mittel PoF Rural PL3	11,8	12,0	11,9
SBR Mittel PoF Rural PL3	12,5	12,8	12,6
L2N Mittel PoF Urban PL3	10,8	10,4	10,6
SBR Mittel PoF Urban PL3	11,0	10,8	10,9

Point of Failure 2 Audio-Beispiele L2N 8 Versuchspersonen SBR 9 Versuchspersonen

Session 7

Onset of Impairment C/N-Mittelwerte L2Normal Mono48kBit Rural und Urban PL3

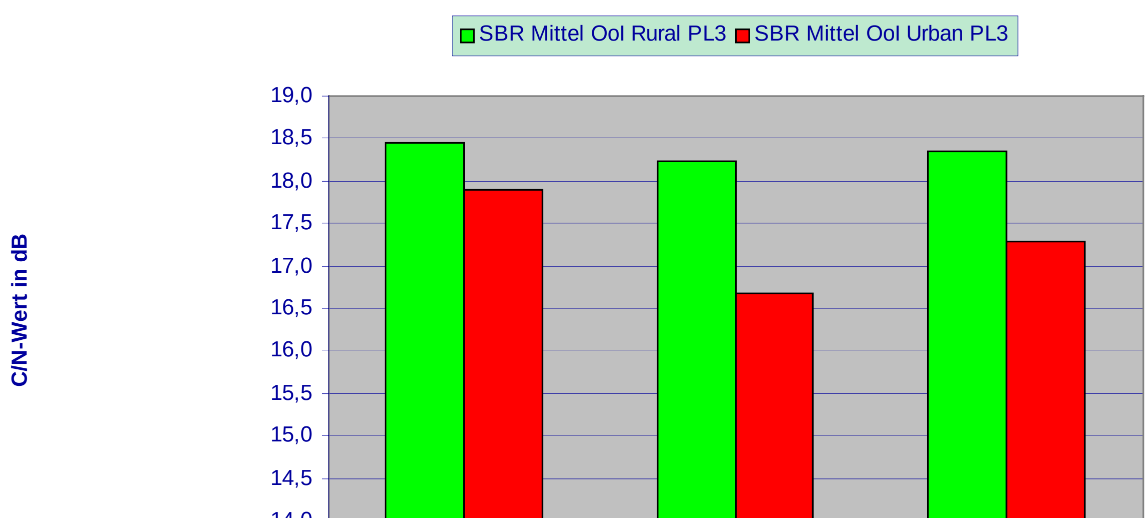


	Frau 8	Mann 9	Sprache
L2N Mittel OoI Rural PL3	17,2	17,8	17,5
L2N Mittel OoI Urban PL3	16,9	16,4	16,7

Onset of Impairment 2 Audio-Beispiele 8 Versuchspersonen

Session 6

Onset of Impairment C/N-Mittelwerte
SBR Mono DualRate LowComp 48kBit Rural und Urban PL3

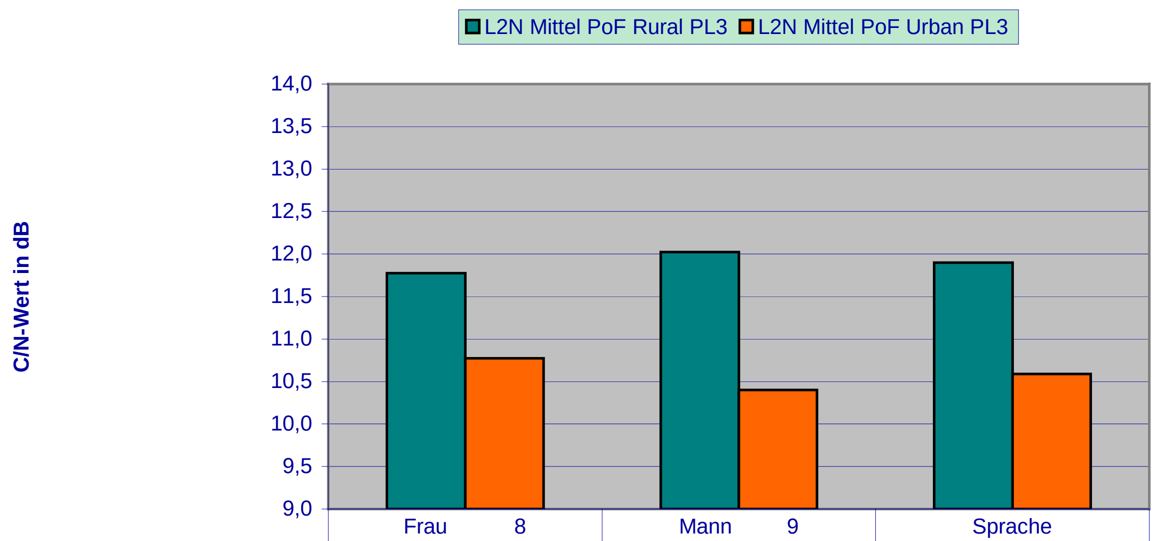


	Frau 8	Mann 9	Sprache
SBR Mittel OoI Rural PL3	18,5	18,2	18,3
SBR Mittel OoI Urban PL3	17,9	16,7	17,3

Onset of Impairment 2 Audio-Beispiele 9 Versuchspersonen

Session 7

Point of Failure C/N-Mittelwerte L2Normal Mono48kBit Rural und Urban PL3

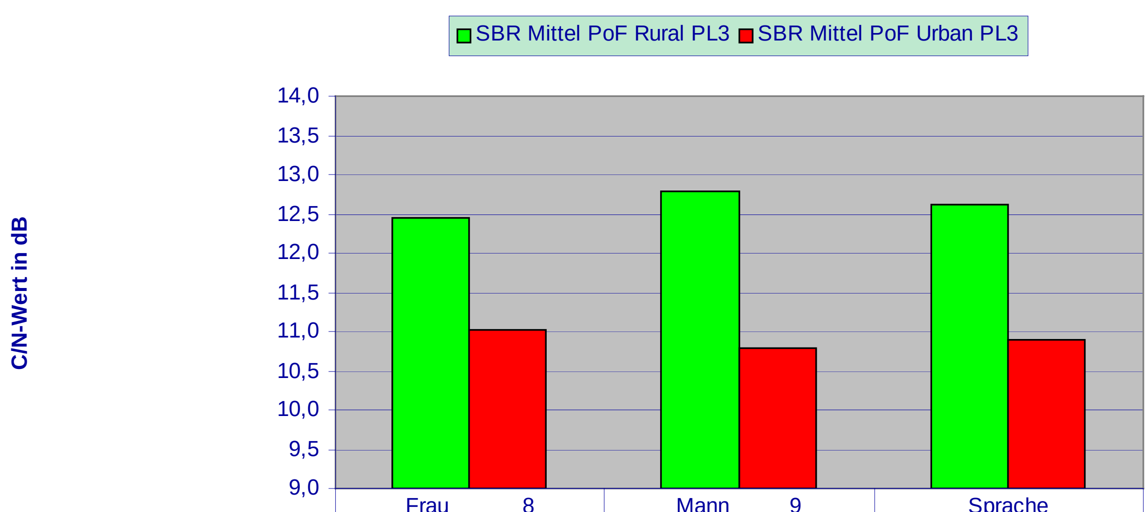


	Frau 8	Mann 9	Sprache
L2N Mittel PoF Rural PL3	11,8	12,0	11,9
L2N Mittel PoF Urban PL3	10,8	10,4	10,6

Point of Failure 2 Audio-Beispiele 8 Versuchspersonen

Session 6

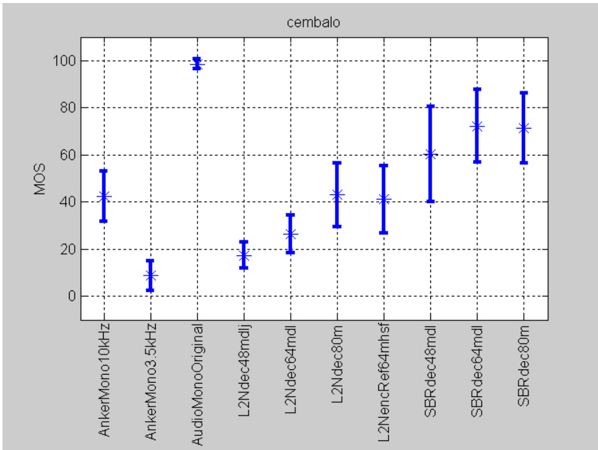
Point of Failure C/N-Mittelwerte
SBR Mono DualRate LowComp 48kBit Rural und Urban PL3

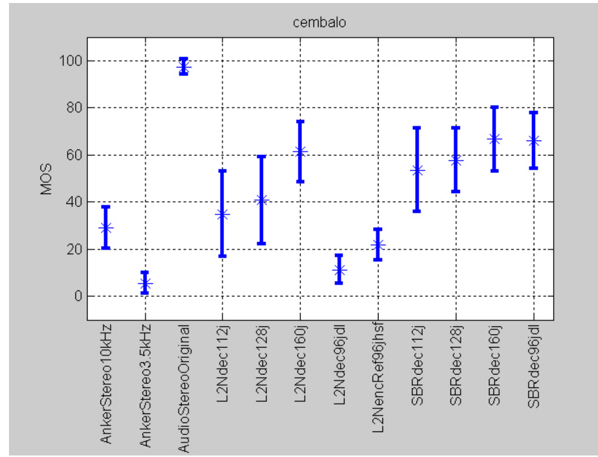


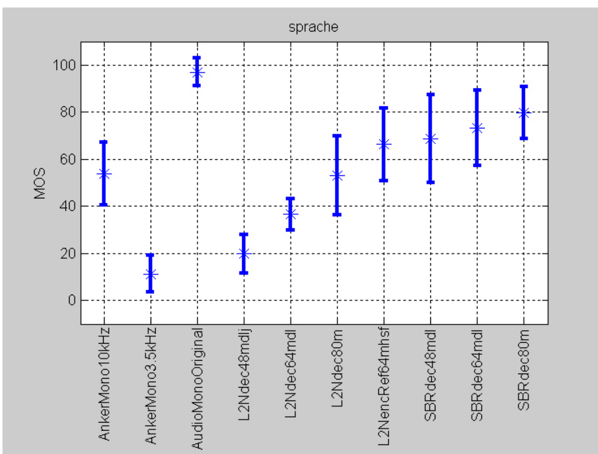
	Frau 8	Mann 9	Sprache
SBR Mittel PoF Rural PL3	12,5	12,8	12,6
SBR Mittel PoF Urban PL3	11,0	10,8	10,9

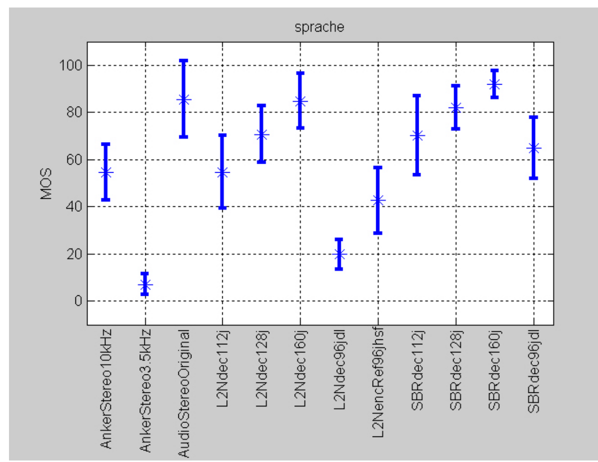
Point of Failure 2 Audio-Beispiele 9 Versuchspersonen

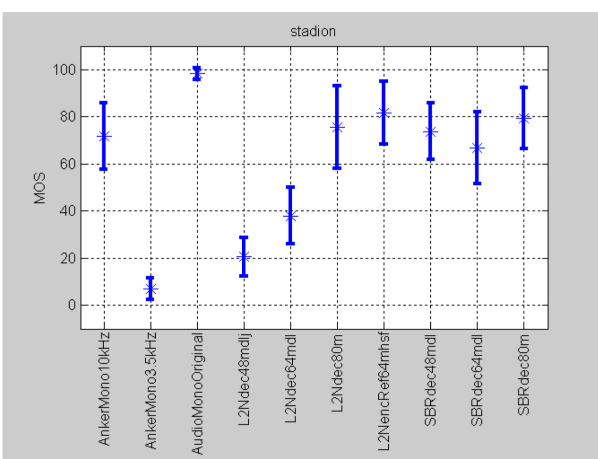
14 Appendix 2: Quality degradation for different audio items

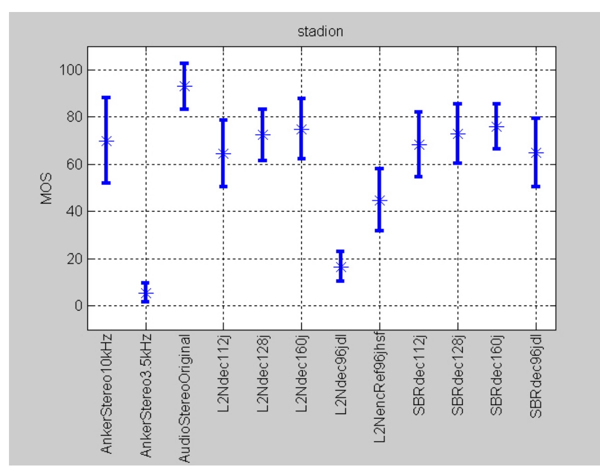


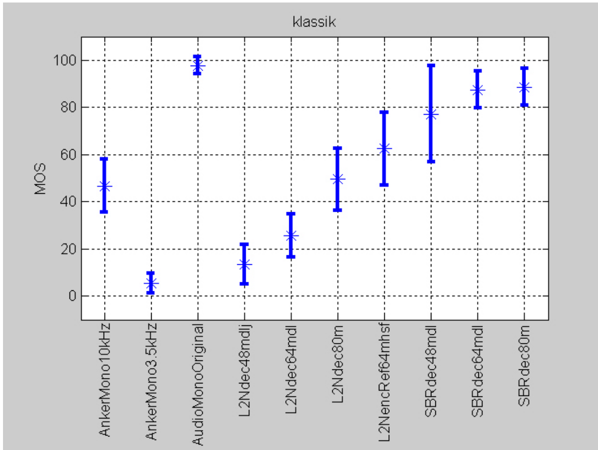


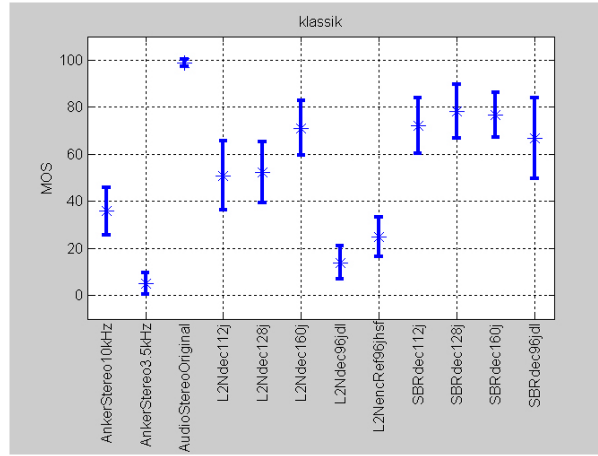









klassik
MOS
100
80
60
40
20
0
AnkerMono10kHz
AnkerMono3.5kHz
AudioMonoOriginal
L2Ndec48mdlj
L2Ndec64mdl
L2Ndec80m
L2NencRef64mhsf
SBRdec48mdl
SBRdec64mdl
SBRdec80m


klassik
MOS
100
80
60
40
20
0
AnkerStereo10kHz
AnkerStereo3.5kHz
AudioStereoOriginal
L2Ndec112j
L2Ndec128j
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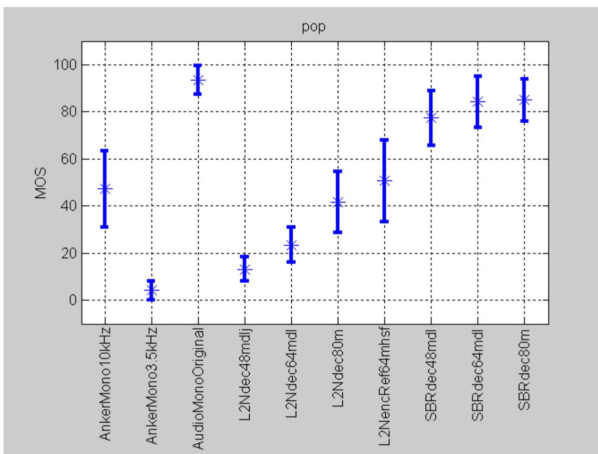
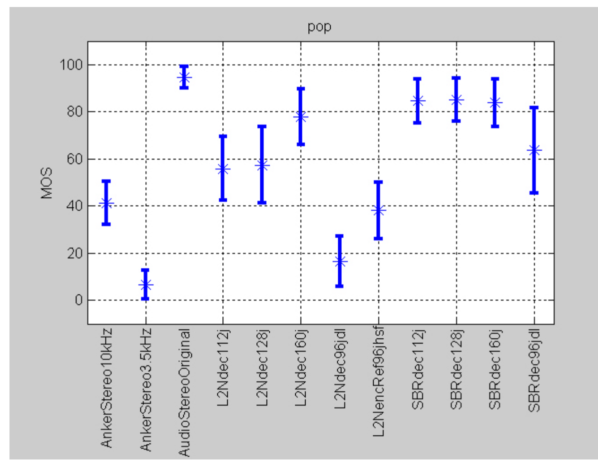
SBRdec160j
SBRdec96jdl


pop
MOS
100
80
60
40
20
0
AnkerMono10kHz
AnkerMono3.5kHz
AudioMonoOriginal
L2Ndec48mdlj
L2Ndec64mdl
L2Ndec80m
L2NencRef64mhsf
SBRdec48mdl
SBRdec64mdl
SBRdec80m


pop
MOS
100
80
60
40
20
0
AnkerStereo10kHz
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AudioStereoOriginal
L2Ndec112j
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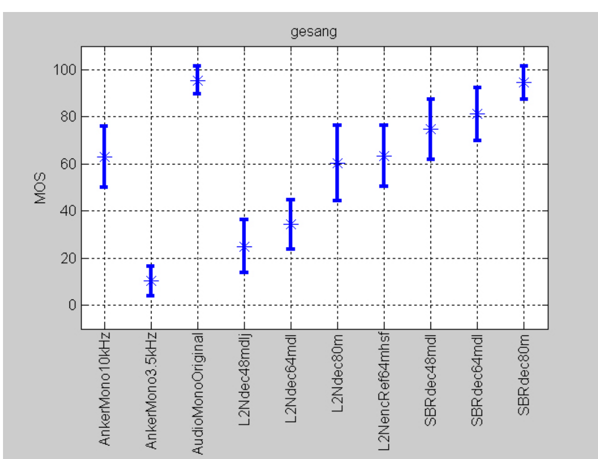
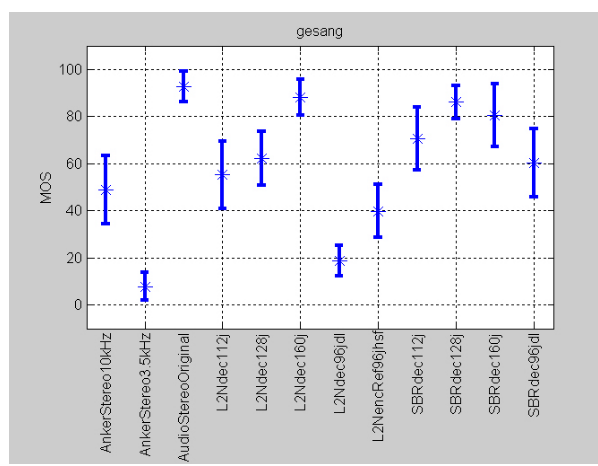

gesang
MOS
100
80
60
40
20
0
AnkerMono10kHz
AnkerMono3.5kHz
AudioMonoOriginal
L2Ndec48mdlj
L2Ndec64mdl
L2Ndec80m
L2NencRef64mhsf
SBRdec48mdl
SBRdec64mdl
SBRdec80m


gesang
MOS
100
80
60
40
20
0
AnkerStereo10kHz
AnkerStereo3.5kHz
AudioStereoOriginal
L2Ndec112j
L2Ndec128j
L2Ndec160j
L2Ndec96jdl
L2NencRef96jhsf
SBRdec112j
SBRdec128j
SBRdec160j
SBRdec96jdl