Diploma Thesis

Audiovisual quality for multimedia content in UMTS networks

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Content

Abstract

Wireless multimedia applications, such as audio and video streaming, becomes reality since the transmission bandwidth increases significant in 3G wireless networks. Together with 3G mobile terminals, cell phones, and other high performance mobile services, the overall perceived audiovisual quality of the end user can be satisfied by specific chosen audio and video codec parameter settings, such as low bitrates, sampling frequencies, low frame sizes, and low frame rates. While for both codec types different low codec settings are possible, it is necessary to find the lowest codec settings to reach the best perceived user quality of a service. In case of mobile audio services, the perceived audio quality, depending on the specific audio codec settings, can be estimated by subjective listener tests or objective quality measurement methods. From another point of view, the information, which should be estimated, is the influence of specific audio codecs and their settings on the perceived audio quality of different coded audio contents. In subjective listener tests, a huge number of different coded audio files with different contents are played to test persons.

This diploma thesis is focused on reference free audio quality estimation for AAC, AMR WB, and AMR NB coded different audio content in mobile environment. The proposed solution provides suitable trade-off between prediction accuracy and computational complexity.

Zusammenfassung

Die signifikante Erhöhung der Übertragungsbandbreite in 3G Wireless Systems ermöglicht die Übertragung von Multimedia Anwendung, wie zum Beispiel Audio und Video Streaming, in bestmöglicher Übertragungsqualität. Zusammen mit den dafür zur Verfügung stehenden 3G mobilen Benutzergeräten (mobile Terminals, Handy's, …) und entsprechend geeigneter Wahl von Audio und Video Codec Einstellungen ist eine bestmögliche Kundenzufriedenheit bezüglich der wahrgenommenen, audiovisuellen Qualität des angebotenen Multimedia Services mittels niedrigen Bitraten erreichbar. Beeinflusst wird dieser kundenspezifische Qualitätseindruck durch die gewählten Einstellungen der Audio und Video Codec Parameter. Mittels subjektiven und objektiven Qualitätsbeurteilungsmethoden ist es möglich, diesen wahrgenommenen Qualitätseindruck des Benutzers zu schätzen. Um diesen Qualitätseindruck zu ermitteln, beurteilen Testhörer in subjektiven Hörtests die Audioqualität einer großen Anzahl an unterschiedlich codierten Audiofiles unterschiedlichen Inhaltes und Audio Codec Einstellungen. Diese subjektiven Testverfahren können durch objektive Qualitätsschätzungsmethoden ersetzt werden, wobei diese den kundenspezifischen Qualitätseindruck entweder mit oder ohne Hilfe von Referenzinformationen durchführen.

Diese Diplomarbeit befasst sich mit der Entwicklung einer objektiven Audioqualitätsschätzungsmethode ohne Referenzinformation für AAC, AMR WB, und AMR NB codierten Audiofiles für mobile Audio Services, wobei die vorgestellte Methode einen geeigneten Kompromiss zwischen Qualitätsschätzungsgenauigkeit und Komplexität des entwickelten Verfahrens darstellt.

Chapter 1

Introduction

Wireless multimedia applications, such as audio and video streaming, becomes reality since the transmission bandwidth increases significant in 3G wireless networks. Together with 3G mobile terminals, cell phones, and other high performance mobile services, the overall perceived audiovisual quality of the end user can be satisfied by specific chosen audio and video codec parameter settings, such as low bitrates, sampling frequencies, low frame sizes, and low frame rates. While for both codec types different low codec settings are possible, it is necessary to find the lowest codec settings to reach the best perceived user quality of a service. In case of mobile audio services, the perceived audio quality, depending on the specific audio codec settings, can be estimated by subjective listener tests or objective quality measurement methods. From another point of view, the information, which should be estimated, is the influence of specific audio codecs and their settings on the perceived audio quality of different coded audio contents. In subjective listener tests, a huge number of different coded audio files with different contents are played to test persons. Those test persons rate the audio quality by different score values, e.g., 1 for bad quality or 5 for excellent quality, if a mean opinion score (MOS) scale is used. Another method to estimate the perceived audio quality of different coded audio files, consisting of different contents, are objective quality measurement methods, realized by mathematic algorithm. Objective quality measurement algorithms are simulating the auditory perception and cognitive part of the human brain for analyzing the perceived audio quality judgement process of human beings. With the output variables of such objective quality measurement models it is possible, to design audio quality estimation metrics for predicting the impression of the perceived audio quality test listener have about a specific coded audio content. Most of the objective quality measurement methods are based on reference information about the original, uncoded audio file. Disadvantage of such reference based quality measurement methods are always the need of reference information.

Goal of this diploma thesis is to develop a reference free audio quality estimation system. The perceived audio quality is estimated for different coded audio contents and audio codec settings bitrate and sampling frequencies. Therefore, the following reference free classification stages were developed:

- audio codec classification stage
- audio codec bitrate and sampling frequency estimation stage
- audio content classification stage
- audio quality estimation stage

Further, the reference free audio quality estimation system is extended by a scene change detection tool to predict the audio codec, audio codec settings, audio content, and audio quality of each audio scene in a video sequence.

Chapter 2

Perceptual audio quality measurement and assessment methods

The perceived audio quality of different coded audio contents, e.g., speech, music, or effect sounds (fx), can be measured by subjective or objective audio quality assessment methods. Objective perceptual audio quality measurement methods, based on mathematic algorithm, are developed to simulate or substitute the situation of subjective perceptual audio quality measurement methods (subjective MOS listener tests), in which test listeners scores the audio quality of different coded audio files with different coded audio codec settings bitrate and sampling frequency. The result of a subjective listener test is called the “mean opinion score” (MOS), an integer value between 1 (bad) and 5 (excellent), which reflects the user opinion of the perceived audio quality of a coded audio file. During a long time, subjective test procedures were the only method of measuring the perceptual audio quality impression of coded audio files. Subjective experiments require a huge number of subjects or test listeners to achieve statistically relevant results, and so, they are very costly and time consuming. Further, the great contrast between evaluation results and perceived audio quality leads to the question, how to interpret the result. For perceptual audio quality measurement, subjective listener tests are not optimal to estimate the perceived audio quality. Therefore, perceptual models, based on mathematic algorithms, can be applied to generate model output variables or parameters for objective audio quality metrics (“objective MOS” or “OMOS”) to predict the perceived audio quality of coded audio content objective audio quality measurement systems). The estimation results of audio quality metrics, consisting of those model output parameters, can be compared with subjective MOS test results or directly with the values of a mean opinion score scale.

In the research area of perceptual model processing, many detailed model output values, such as frequency spectrum components, dynamically measured bandwidth, distortions, or modulation will be generated and reported for making this technology universally applicable.

Those objective quality assessment methods stand in strong relation to the behaviour of human perception and human judgement. To predict the perceived audio quality of a coded audio file of a mobile audio service (audio streaming), it is necessary to find optimal relations between parameter, which can be measured, (transmission errors, noise, distortion and losses due to low bit-rate coding and packet transmission) and the human quality perception process. Once, such a relation is found, audio, video, or audiovisual quality estimation metrics, based on reference information or not, can be designed.

Until now, many different quality evaluation metrics for low bitrate applications have been proposed [1-5]. The basic metrics for non-reference free perceptual audio quality estimation are including “objective” criterions, such as objective difference grades or mean-square error based criterions [6], and results from subjective listener tests to estimate the user perceived audio quality. While in subjective quality assessment methods test persons rate the quality of a service or the quality of coded multimedia content, objective quality measurement or assessment methods try to predict those user perceptual quality impressions using mathematic algorithm. Those objective measurement or assessment algorithms estimate the perceived audio quality using reference information (reference based objective assessment methods) or not (reference free objective assessment methods).

2.1 Objective perceptual audio quality measurement and assessment methods

Objective audio quality assessment methods try to simulate cost- and time expensive subjective audio quality measurement methods (subjective listener tests) by perceptual measurement algorithms. The basic structure of objective perceptual audio quality measurement systems can be divided into a perceptual model stage, a feature extraction stage, and a cognitive model stage for the objective measurement, as described in more details in section 2.1.1. Objective audio quality measurement algorithms or systems can be divided into two main groups: reference free and reference based objective perceptual measurement systems. Input of reference based perceptual measurement systems are the coded, degraded audio file and its original, uncoded version as reference source. Such perceptual measurement algorithm are always reference based, that means, that there is always a need for a reference

information to calculate the perceptual Measurement Output Variables (MOV's), which are further used as the parameter of audio quality estimation metrics. While the coefficients of those parameter are always audio codec and audio content specific, it is possible to choose automatically the audio codec and audio content specific parameter coefficients for each audio codec and audio content by using an audio codec and audio content classification stage.

The development of an effective objective perceptual quality measurement or assessment method for a specific type of telecommunication signal, such as audio or speech, is a significant research problem. Several objective quality assessment methods have already been developed in recent years and those methods may be applied directly on a perceptual model output, which simulates the human auditory system and cognitive behaviour of human beings [7-14]. Further, the output of a good objective quality measurement method should have a high correlation with many different subjective experiments. The closeness of the fit between the results of an audio quality estimation metric, consisting of objective quality measurement model or system output parameters, and the results of subjective test listener scores can be measured by calculating a correlation factor or coefficient (cf. section 5.6.1).

As described above, objective measurement methods try to simulate the human perception and human cognition behaviour using psychoacoustic models, based on psychoacoustic phenomens, as investigated, e.g., in [15], [16], [17]. The most advanced objective perceptual quality assessment methods may be found in the areas of audio and speech, while for those telecommunication signals psycho-acoustic effects, known from masking experiments, are differing in a significant way. For wideband audio signals, the PEAQ (Perceptual Evaluation of Audio Quality) method has been developed and recommended by the ITU-R Rec. Bs. 1387 [7]. PEAQ was developed originally as an automated method to evaluate the perceptual quality of different wideband audio codecs. Such codecs are used to sample wideband audio signals and compressing the bit rate requirements. By applying the PEAQ algorithm to the individual packet streams within the network performance model (e.g., the packet-switched mobile networks, such as 3GP), it is possible to obtain objective perceptual quality output measurement indicators for the investigated audio stream or audio file, which is analyzed by the model. For speech signals, several objective perceptual quality assessment methods have been developed, e.g., PAMS (Perceptual Analysis Measurement System) [8], [9], PSQM (Perceptual Speech Quality Measurement) [10], and PESQ (Perceptual Evaluation of Speech

Quality), recommended by the ITU standard P.862 [11], or PEAQ (Perceptual Evaluation of Audio Quality) [7] as objective perceptual audio quality measurement systems for coded audio content. All of those methods are using natural or artificial speech signals as input reference to provide a perceptual based output for further quality scores. The basic structure of those reference based objective measurement algorithms or systems is shown in Fig. 2.1 [12].

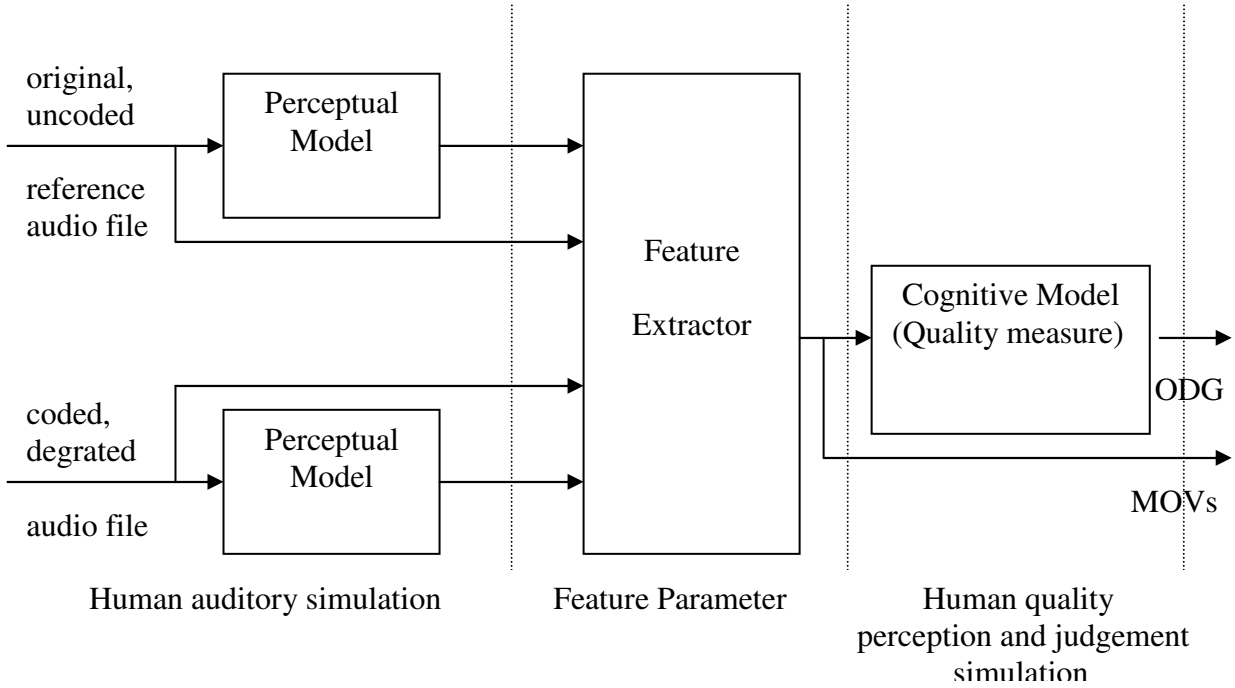


Figure 2.1: Basis structure and main components of objective measurement algorithm [12].

This basic structure is non reference free, consisting of two input audio signals, the uncoded, original audio signal or audio file as reference, and the coded audio signal. The three main components in reference based objective measurement algorithms are:

1.) Perceptual model: models the peripheral ear to simulate the human auditory system

2.) Feature extraction: comparison of the outputs of the ear model for modelling the audible distortion presented in the coded, degraded audio file. Those extracted feature parameter are called Model Output Variables (“MOVs”), and are used, together with

objective difference grade parameters (ODG), as the parameters in audio quality estimation metrics.

3.) Deriving of a quality measure: consisting of a single number that indicates the audibility of the distortions presents in the degraded audio file. This is done by simulating the cognitive part of the human auditory system through further processings of the model output variables MOVs.

Reference based perceptual models, as shown in Fig.2.1, use the undegraded, original audio file and its coded version as input to calculate the audio codec and audio content specific objective difference grade and model output variables. For predicting the audio quality of different coded audio content automatically, the perceptual model should be extended with an audio codec and audio content classification stage, to choose the codec and content specific audio quality estimation metric and its coefficients. Examples for audio quality estimation metrics for AMR and AAC coded speech content are given in equation (2.1) and (2.2), while equation (2.3) gives an example for an audio quality estimation metric, suitable to predict the perceived audio quality of AAC coded music content [1]:

AMR coded speech content: $MOSA = - 6.996 \cdot AD^2 + 10.95 \cdot AD + 1.165$ (2.1)

AAC coded speech content: $MOSA = - 6.996 \cdot AD^2 + 10.95 \cdot AD + 0.370$ (2.2)

AMR / AAC coded music content:

$$MOSA = - 3.1717 + 4.8809/IFD + 0.3562 \cdot A\_ind + 0.0786 \cdot D\_ind \quad (2.3)$$

The metrics in equation (2.1) and (2.2) are based on the model output parameter auditory distance AD from the Measuring Normalizing Block Technic for Objective Estimation of Perceived Speech Quality [13], measuring dissimilarities between the original and degraded, coded audio file.

The audio quality estimation metric parameters in equation (2.3) are based on the PESQ (Perceptual Evaluation of Speech Quality) [11] model output parameters integrated frequency distance IFD and the disturbance indicators A_ind and D_ind. Further, an example for a perceptual speech quality metric for waveform codecs, CELP / hybrid codecs, and mobile codecs / systems, based on model output variables symmetric and asymmetric disturbance indicators (d_{SYM}, d_{ASYM}) of the PESQ system, is presented in equation (2.4) [14]:

$$PESQ_{MOS} = 4.5 - 0.1 \cdot d_{SYM} - 0.0309 \cdot d_{ASYM} \quad (2.4)$$

The following section gives an overview of the international standardized perceptual audio measurement methods mentioned in the section above:

1996 PSQM [10] : Perceptual Speech Quality Measure (intrusive), developed originally 1996 by KPN Research (Netherlands) and is now specified in the ITU-T recommandation P.861 [10] PSQM was the first method based on psychoacoustic measuring for predicting listening quality. While the use of PSQM is essentially limited to assessing the quality of continous speech signals, the intrusive Avanced Speech Quality Measure, developed 1998 also by KPN Research. PSQM+ is more suitable for packet speech measurements. Further, PSQM+ improves the time- alignment of the signals to be compared and also how silence periods and packet dropouts are taken into account in evaluating subjective quality. In comparison to PSQM+, it can be seen as a kind of basic “core” model with no gain- or time-alignment for signal preprocessing.

1998 PEAQ [7] : Perceptual Evaluation of Audio Quality according to ITU-R recommendation BS.1387, available as a basic and advanced model.

1999 MNB [13] : MNB is an Objective Estimation of Perceived Speech Quality using a measuring normalizing block technique (MNB) and was developed by Stephen Voran from NTIA as an appendix to recommendation P.861. Since PSQM is limited to higher bit rate speech codecs operating over error-free channels, it is not suitable for an objective measurement of the

perceived quality of highly compressed digital speech with bit errors or frame erasures.

2000 PAMS [9] : Perceptual Analysis Measurement System, developed by the British Telecom and was the first method to provide robust results for packet voice signals

2001 PESQ [11] : Perceptual speech quality measure (intrusive), developed 2001 in a collaboration between British Telecom and KPN Research and is the new ITU-T Recommandation P.862 [11], [14]. It combines PSQM with PAMS and is optimized for VoIP and hybrid end-to-end applications. PESQ is the preferred method for measuring the perceptual quality of packet speech signals.

Objective perceptual quality assessment methods are also being developed for video signals, in particular by the Video Quality Experts Group (VQEG) [18], [19].

2.2 Subjective perceptual audio quality measurement and assessment methods

Subjective mean opinion score (MOS) listener tests are necessary to get an impression, how human beings are classifying the audio quality of coded audio content. With this information, it is possible to design audio quality estimation metrics. In a subjective MOS test scenario, different coded audio files are randomly played to the test listener, which classifies the perceived audio quality impression of each coded audio file by using values of a MOS scale in the range (1..5).

To get an impression of the perceived audio quality of different coded audio contents, in the subjective MOS listener tests, coded audio test files were randomly numbered by hand at an i-tune playlist in that way, that the audio content and the audio codec were different for each MOS rating test. The playbacks of the audio files were done using i-tunes, and the audio files

of different content (speech, fx sounds, classic music, other music) were coded with AAC, AMR WB, and AMR NB audio / speech codecs, using different audio codec settings bitrate and sampling frequencies, whereas the sub audio category other music consists of non classic music styles like pop, rock, trance, and techno. Table 2.1 gives an overview of the audio codecs, audio codec settings, and audio content types for the subjective MOS listener test scenario:

Audio codec	bitrate [kbps]	sampling frequency [kHz]	audio content
AAC	8	8	speech, non speech
AAC	16	16	speech, non speech
AAC	20	16	speech, non speech
AAC	24	22.05	speech, non speech
AMR WB	6.6	16	speech, non speech
AMR WB	8.85	16	speech, non speech
AMR WB	12.65	16	speech, non speech
AMR WB	15.85	16	speech, non speech
AMR NB	4.75	8	speech
AMR NB	7.95	8	speech
AMR NB	12.2	8	speech

Table 2.1: Audio codecs, audio codec settings, and audio content of the subjective MOS listener test scenario.

The different coded audio content listener test setup consists of a speech file, an effect sound (fx sound), a dance / techno file, and a classic music file, representing music with strong and easy rhythm. Finally, 35 different coded audio files were used for the subjective MOS listener tests. The subjective MOS listener tests consists of two rounds, in which the test listeners had classified the perceived audio quality by using a MOS scale value between 1 (bad) and 5 (excellent), and each run has taken 12 minutes. In a training session, from each audio content type the AAC 24kbps coded audio file version was chosen and played to the test person with

individual choosen playback volume level. AAC at 24kbps with 22.05kHz was chosen in the training phase while this audio codec gives the best overall quality in the test scenario setup, in comparison to the other used audio codecs and audio bitrates. So, the perceived audio quality of AAC coded audio content at 24kbps can be seen as the upper bound of the perceived audio quality / MOS scale (perceived audio quality reference value) and the perceived audio quality of all other audio codecs were rated by the test listeners in relation to this maximum perceived audio quality audio codec bound.

2.2.1 Subjective MOS listener test results for different coded audio files

2.2.1.1 Test results for different coded speech files

The subjective MOS listener test result for the speech file speech_stadt.wav, coded with AAC, AMR WB, and AMR NB at different bitrates and sampling frequencies, are shown in Fig.2.2:

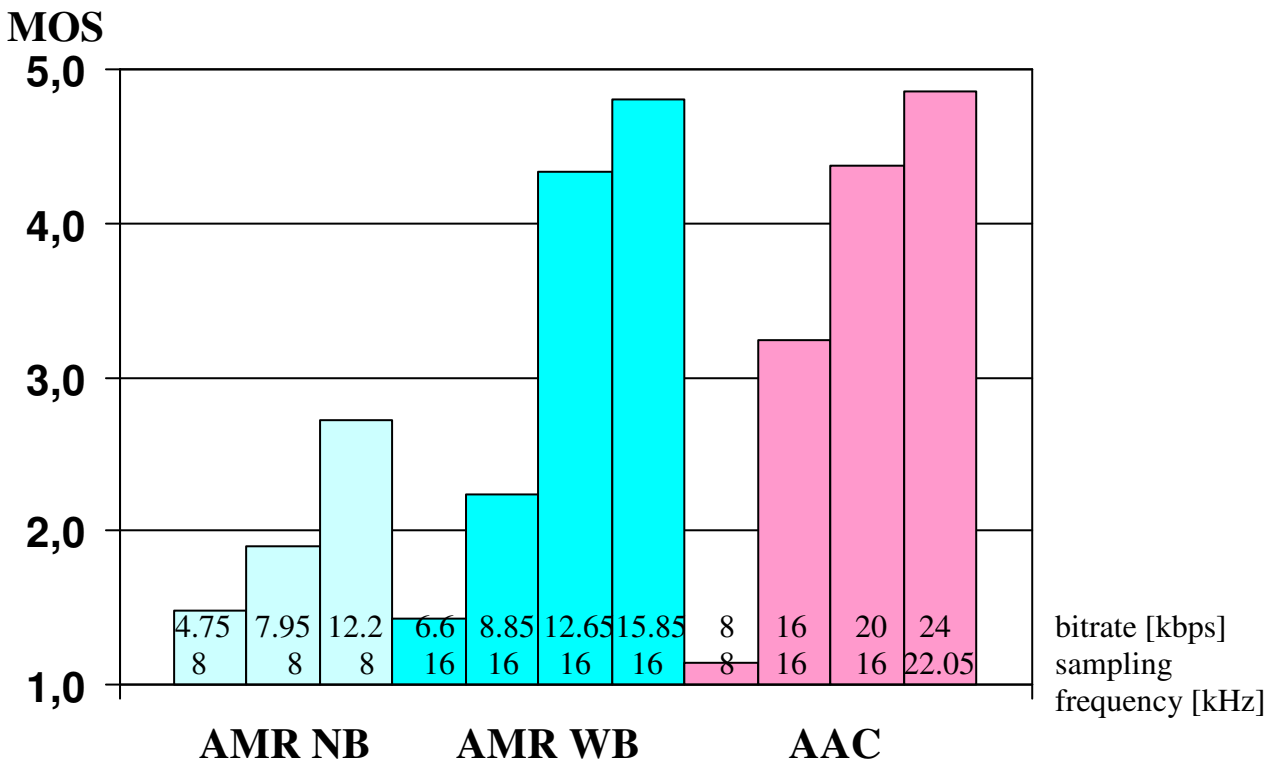


Figure 2.2: MOS test results for AMR WB, AMR NB, and AAC coded speech content.

The relations between audio codecs, audio codec settings and subjective MOS test results for the test file speech_stadt.wav are shown in Table 2.2:

audio content	audio codec settings	audio quality rated as
Speech	AMR NB 4.75kbps, 8kHz	bad 1
Speech	AMR NB 7.95kbps, 8kHz	poor 2
Speech	AAC 8kbps, 8kHz	bad 1
Speech	AMR NB 12.2kbps, 8kHz	fair 3
Speech	AMR WB 6.6kbps, 16kHz	bad 1
Speech	AMR WB 8.85kbps, 16kHz	poor 2
Speech	AMR WB 12.65kbps, 16kHz	good 4
Speech	AMR WB 15.85kbps, 16kHz	excellent 5
Speech	AAC 16kbps, 16kHz	fair 3
Speech	AAC 20kbps, 16kHz	good 4
Speech	AAC 24kbps, 22.05kHz	excellent 5

Table 2.2: Audio codecs, audio codec settings bitrate and sampling frequency, and audio content of the subjective MOS listener test scenario, speech content.

Those results for AAC, AMR WB, and AMR NB coded speech content show, that the perceived audio quality for AMR WB coded speech content is much more better than for AMR NB coded speech content. Comparing the mean MOS test result for AMR NB 7.95kbps with AAC 8kbps, both sampled at 8kHz, the perceived audio quality for AMR NB coded speech content is better rated as for the same speech content coded with AAC. The mean value of the MOS for AMR WB 12.65kbps is significant better than the mean value of the MOS for AMR NB 12.2kbps: rated as 4 instead of 2.5. Further, it is possible to reach the same MOS value using AMR WB at 12.65kbps instead of AAC 20kbps, and AMR WB 15.85kbps instead of AAC 24kbps. This means, comparing to previous work [1-3], where the overall best MOS value in the audio video scenario is reached by AAC 24kbps (even better as AMR NB), that it is possible to reach the best overall perceived audio quality for speech with audio codec settings AMR WB 15.85kbps and sampling frequencies 16kHz. Taking a look at

the bad MOS test results for AMR WB and AMR NB for speech content, it seems not to be clear, why AMR NB at 4.75kbps, 8kHz should be perceived better or similar to AMR WB at 6.6kbps, 16kHz.

Listening to the sound sample, it can be heard, that the AMR WB coded speech file sounds "brighter" than the AMR NB coded version, caused by a sampling rate 16kHz for AMR WB instead of the AMR NB sampling frequency 8kHz. It seems that the test listeners are more sensitive about distortions in "brighter" audio files than for distortions in "smoother" audio files. Maybe this can be compared with the effect of the human visual system in relation to the field of television technic, where errors are represented by the color black in fact, that the human eye is not so sensitive to black than white. Another reason why the speech codec AMR WB at lower bitrates 6.6kbps and 8.85kbps is rated as bad, can be seen in the maximum bound of the perceived audio quality of their AAC 24kbps versions and sampling frequency 22.05kHz. Further, in relation to the good MOS value of AMR WB at 12.65kbps and 15.85kbps, and in relation of AAC 20kbps and AAC 24kbps, the test persons perceived a good threshold for what they rated as good or bad.

2.2.1.2 Test results for different coded other music

Test results from the subjective MOS listener test for the pop / techno test file other_music_FA.wav, coded with AAC, AMR WB, AMR NB at different bitrates and sampling frequencies, are shown in Fig.2.3:

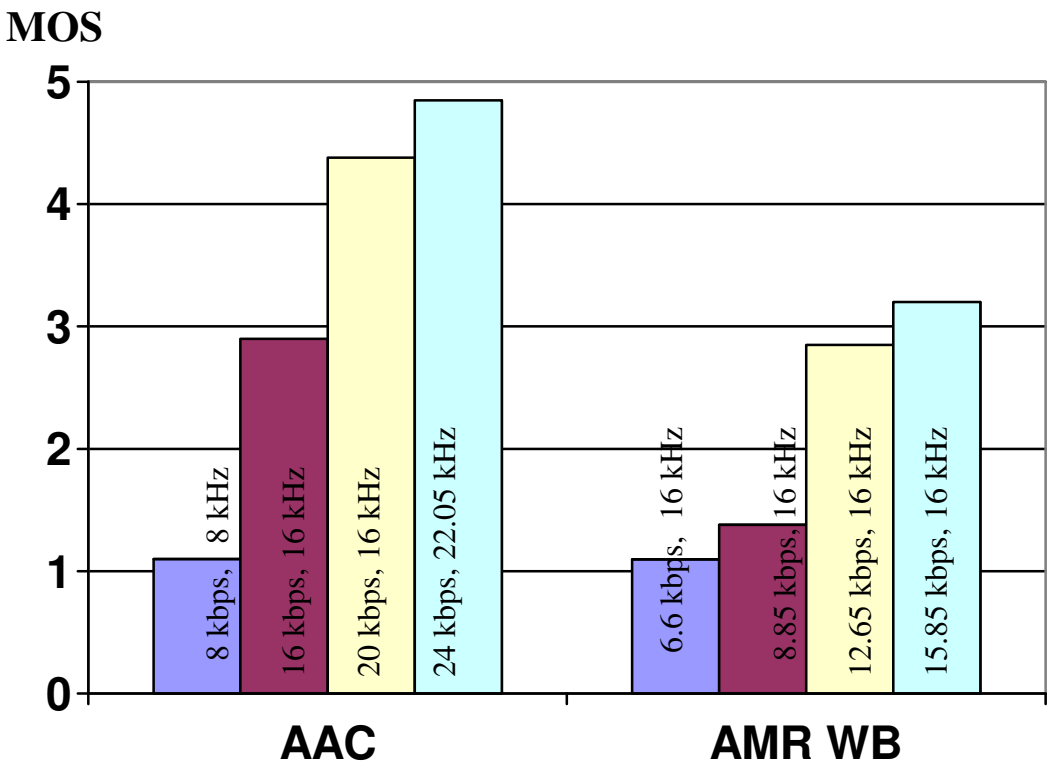


Figure 2.3: MOS test result for AAC and AMR WB coded other music.

The different colours of the bars in Fig.2.3 are containing no color codes or relations between the audio codecs and bitrates; they are just for illustrations and are containing no colour code. The relations between audio codec, audio codec settings bitrate and sampling frequency, and subjective MOS test results for the test file other_music_FA.wav content are shown in Table 2.3:

audio content	audio codec settings	audio quality rated as	
other music	AAC 8kbps, 8kHz	bad	1
other music	AMR WB 6.6kbps, 16kHz	bad	1
other music	AMR WB 8.85kbps, 16kHz	bad	1
other music	AAC 16kbps, 16kHz	fair	3
other music	AMR WB 12.65kbps, 16kHz	fair	3
other music	AMR WB 15.85kbps, 16kHz	fair	3
other music	AAC 20kbps, 16kHz	good	4
other music	AAC 24kbps, 22.05kHz	excellent	5

Table 2.3: Audio codecs, audio codec settings bitrate and sampling frequency, and audio content of the subjective MOS listener test scenario, other music.

Those test results for AAC and AMR WB coded other music show, that AMR WB is not a suitable low bitrate audio codec for other music content in relation to high audio quality at low bitrates, and the results for AAC coded music content are similar to previous works [1- 3]. The most suitable audio codec for other music in relation to the perceived audio quality is AAC with codec settings 24kbps and sampling frequency 22.05kHz.

2.2.1.3 Test results for different coded classic music

Test results from the subjective MOS for the different coded classic music test file classic_music_haydn.wav for different audio codecs and audio codec settings are shown in Fig.2.4:

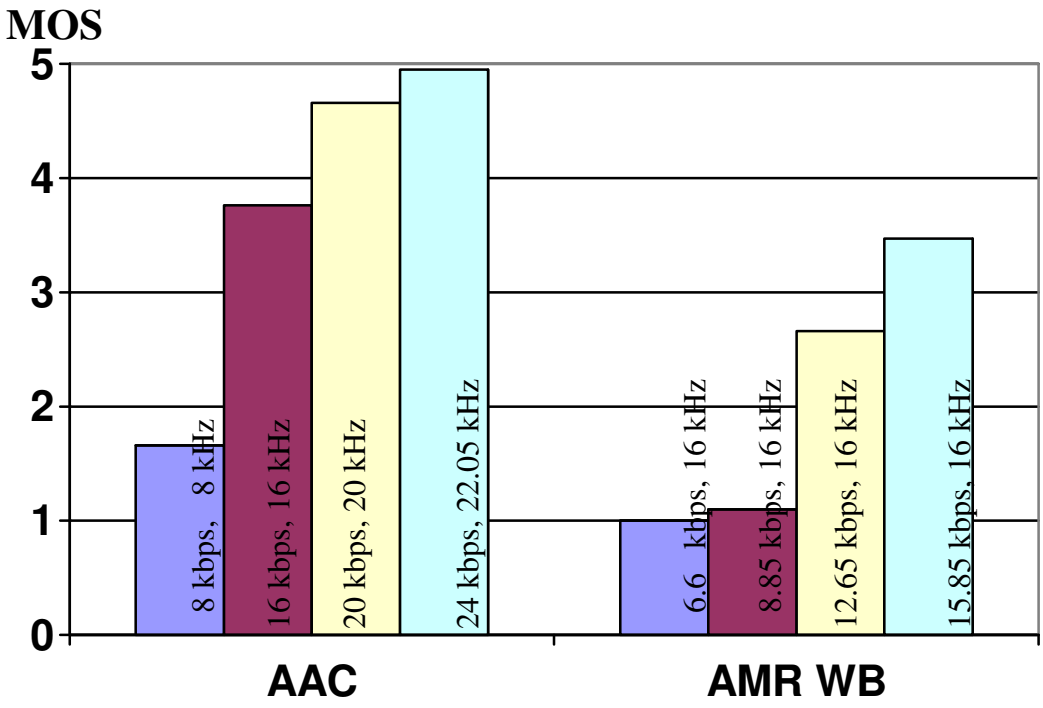


Figure 2.4: MOS test results for AAC and AMR WB coded classic music.

Similar to Fig.2.3, there is no relation between the different colours of the bars in Fig.2.4 and the audio codecs and bitrates; they are just for illustrations and are containing no colour code. The relations between audio codec, audio codec settings and subjective MOS test results for classic music audio test file classic_music_haydn.wav are shown in Table 2.4:

audio content	audio codec settings	audio quality rated as	
Classic music	AAC 8kbps, 8kHz	good	2
Classic music	AMR WB 6.6kbps, 16kHz	bad	1
Classic music	AMR WB 8.85kbps, 16kHz	bad	1
Classic music	AMR WB 12.65kbps, 16kHz	fair	3
Classic music	AMR WB 15.85kbps, 16kHz	fair	3
Classic music	AAC 16kbps, 16kHz	good	4
Classic music	AAC 20kbps, 16kHz	excellent	5
Classic music	AAC 24kbps, 22.05kHz	excellent	5

Table 2.4: Audio codecs, audio codec settings, and audio content of the subjective MOS listener test scenario, classic music.

Similar to the results of AAC and AMR WB coded other music, AMR WB is not a suitable low bitrate audio codec for classic music. But, comparing the bitrate MOS relation of AAC 16kbps coded classic music with those of AAC 16kbps coded other music, the test results show, that there is a significant better MOS value for classic music coded AAC 16kbps (MOS equal 4, good) comparing to other music (MOS equal 3, fair). As Table 2.4 shows, AAC coded classic music with codec settings 20kbps and sampling frequency 16kHz is rated equal AAC coded classic music with codec settings 24kbps and sampling frequency 22.05kHz. So, it is possible to reach the same perceived audio quality for AAC codec classic music wih lower bitrate and sampling frequency.

2.2.1.4 Test results for different coded effect sounds

The results of the subjective MOS listener test for the different coded test file fx_sound_stadion.wav are shown in Fig.2.5, in which the different colours of the bars do not contain any color code, and the audio codec, audio codec settings and subjective MOS test results are given in Table 2.5:

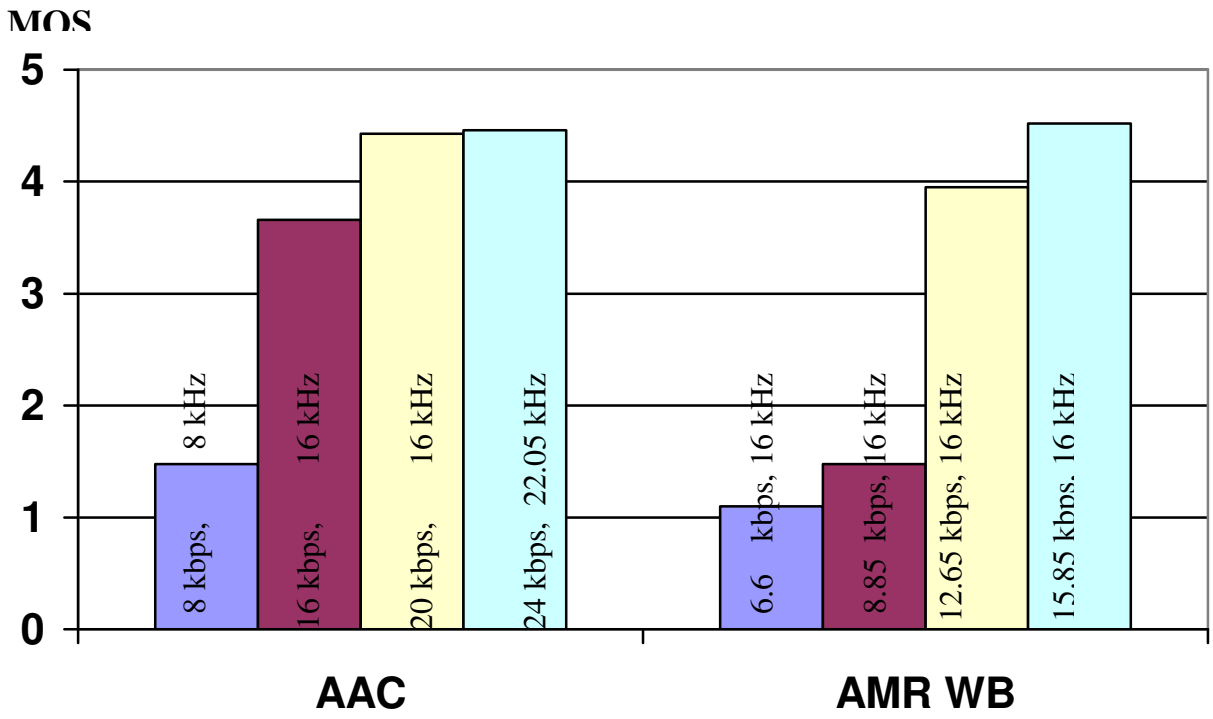


Figure 2.5: MOS test results for AAC and AMR WB coded fx sounds.

The relations between audio codec, audio codec settings and subjective MOS test results for classic music audio test file fx_sound_stadion.wav are shown in Table 2.5:

audio content	audio codec settings	audio quality rated as	
ambient, fx sounds	AAC 8kbps, 8kHz	bad	1
ambient, fx sounds	AMR WB 6.6kbps, 16kHz	bad	1
ambient, fx sounds	AMR WB 8.85kbps, 16kHz	bad	1
ambient, fx sounds	AAC 16kbps, 16kHz	good	4
ambient, fx sounds	AAC 20kbps, 16kHz	good	4
ambient, fx sounds	AAC 24kbps, 22.05kHz	excellent	5
ambient, fx sounds	AMR WB 12.65kbps, 16kHz	good	4
ambient, fx sounds	AMR WB 15.85kbps, 16kHz	excellent	5

Table 2.5: Audio codecs, audio codec settings bitrate and sampling frequency, and audio content of the subjective MOS listener test scenario, fx sound.

Those test results show, that the perceived audio quality of an ambient sound, coded with AMR WB 15.85kbps, 16kHz are rated equal its AAC 24kbps version, sampled at 22.05kHz. So, for the case of ambient fx sounds, it is possible to reach the same perceived audio quality at lower bitrates and sampling frequencies. Further, the same ambient content coded with AMR WB 12.65kbps, sampled at 16kHz is equal rated as its AAC 16kbps and AAC 20kbps versions, both sampled at 16kHz.

2.3 Conclusion: subjective MOS test results

Table 2.6 resumes the most suitable audio codecs and audio codec settings for specific audio contents, as results from the subjective MOS listener tests:

audio content	audio codec	bitrate [kbps]	sampling frequency [kHz]
Speech	AMR WB	15.85	16
fx sound	AMR WB	15.85	16
classic music	AAC	24	22.05
other music	AAC	24	22.05

Table 2.6: Most suitable audio codecs and their settings for specific audio content types.

Further, the results from the subjective MOS listener tests show, why it is necessary to develop one specific audio quality estimation metric for each audio codec, where the relation to the coded audio content is given by the metric coefficients of the metric parameter, given by the model output variables and objective difference grades of objective quality measurement system.

Chapter 3

Audio content classification

3.1. Overview

For audio quality estimation metrics and audiovisual quality estimation metrics to predict the audiovisual quality for mobile streaming services it is necessary to develop an audio content estimator to identify the audio content as speech or non speech, whereas non speech consists of the audio sub categories effect sounds, classic music, other music. With such an estimator it is possible to design automatic audiovisual metrics for audio and video quality estimation / prediction. Such quality metrics consists of different coefficients for each kind of audio content.

Related works have shown [1-5], that in case of audio it is necessary to evaluate two different audio quality metrics, one for speech and one for non speech sequences, more exactly, one audio quality metric for each kind of audio content (e.g., speech, non speech, different kind of music styles, ambient / fx sounds). While non speech sequences, like music, sound effects, noises, speech with background music or environmental sounds contain much more information as speech, their main application can be found in cinema trailers, documentations, advertisement or video clips (singer with background music). Based on this background, a video stream or clip can be classified by its audio content (see chapter 6).

Existing works in the area of audio content classification presents different classification methods, see [20 - 31] for more details. In [20], [22], [24] audio content classification is based on signal characteristics in the time domain, in the frequency domain, in the time-frequency domain and in the coefficient domain. All of them are using feature parameter extraction units to classify sound signals by their individual characteristics. A good overview of parameters which are usable for feature extraction is given in [20], [21], [22], [24], [25]. While in most of the applications, only one parameter is not enough to design an optimal content classifier, the

combination of two or more parameter to a parameter mix (vector) in relation to the application, as described, e.g., in [20], [21], [23] and [25], leads to acceptable classification results. On the other hand, not all possible parameter which can be extracted from an audio file are necessary for audio content classification, which is demonstrated in [21]. Further parameter optimizations will lead to a much more lower complexity of the final audio quality metrics. An example for the whole process of audio content classification by feature extraction, feature vector design based on the mix of four chosen parameters and the classification in subgroups is presented in [22]. For the final audio quality metric, an audio content estimator for speech, music and speech with music is as important as an audio content classification only for music genres and so, content classification methods, which are based on the main difference in the nature of speech and non speech / music signals are necessary. As mentioned in the previous chapter, there will be one audio quality metric designed for each audio content type and so the number of different audio quality metrics corresponds to the number of different chosen audio content subgenres. The periodicity in an audio content is one of the most suitable characteristic for speech and music classification. This means, that in all kind of music there can be found periodic patterns, given by the natural kind in the rhythm [26], [31] of the audio content. Otherwise, such periodic patterns cannot be found in speech content. The behaviour of speech signals is more random-like. Those characteristics can be shown in the time domain and in the frequency domain:

- Time domain:
  An indicator for periodic structures in a music signal can be found in the zero-crossings of the signal, see [20], [22]. Here, the zero-crossings of a music signal will appear nearly exactly after the same time-interval, variating only in a very small range.

- Frequency domain:
  In the frequency domain, the periodic pattern of music can be presented in the spectral frequency density over the time, similar to [20], [22], and [25]. Such a mean power pattern will show no periodic structure in the spectral frequency density for speech.

The first method is realized by analyzing the zero-crossing rate [20], which is the standard deviation of the zero-crossings in all frames, divided by the frame length, while the second method uses several subband energy ratios [24] to classify speech from music. So, two kinds of estimation methods are investigated and compared: one in the time domain and one in the frequency domain. Both of them are very fast and easy to implement in existing digital audio signal processing programs. The estimators were proofed by the same test setup of audio files to enable a comparison between them. This test setups contains different audio files from different sources (CD or video) and have different lengths and contents. Some of them are extracted from cinema trailers by hand for representing very short audio cuts to simulate the further implementation of the estimators in video-cut depended audio content classification systems. Further investigations of the estimators proof their suitability for different coded audio contents. They should identify AAC or AMR WB / AMR NB coded speech with background music, fx sound as music (non speech sequences), and AAC or AMR WB / AMR NB coded speech as speech content.

3.2. Audio content classification based on zero-crossing rate estimator

3.2.1 General aspects

The first audio content classification method is realized by time domain characteristics of the audio signal, so the original sound can be analysed without further transformations. This method is similar as described in [20], [21], [22], [23]. According to [22], the audio file is separated in frames of 150 sample values with 50% overlapping. Then, the number of zero-crossings of each frame is calculated. A zero crossing occurs when the signal changes its phase and can be detected when consecutive samples have different signs [22]. As described in [24], the zero-crossing ratio is calculated by the number of time-domain zero-crossings divided by the total number of samples in a frame [28]:

$$ZCR = 0.5 \cdot (N-1) \cdot \sum_{m=1}^{N-1} | \, sgn [x (m+1)] - sgn [x (m)] \, | \qquad (3.1)$$

where

sgn [·] … sign function
x(m) … discrete audio signal, m =1 … N.
N … frame length
0.5 … frame overlapping factor (50%)

While different audio sources have special characteristic in their zero-crossing rates, which is illustrated in [20] and [22], this characteristics can be used as an estimator to separate uncoded audio content classes speech, music, and all different styles of music and speech with background music / noise. Speech with background music / noise can be further classified as music.

Finally, to find a threshold value, the standard deviation of the zero crossing rates of all frames, is a suitable indicator for classification. This threshold was found by analyzing the audio test setup (see chapter 6.2.2) results at the value of 0.09 in case of uncoded speech or non speech content.

3.2.2 Audio content classification for different audio content

As shown in related works [20], [22], and [29], the zero-crossing rates of different kinds of audio contents follow typical characteristics for each content type. Typical speech characteristics in relation to the zero crossing amplitudes are shown in Fig.3.7: high peaks of the zero-crossing amplitudes over a relative low and stable line, which can be seen as a kind of “baseline”, in relation to [20]. This results in a wide range (represented by the peaks and troughs, caused by unvoiced and voiced components) and a large variance of the amplitudes. The so called “baseline” is better demonstrated in case of music.

Fig.3.2a shows the zero-crossing-rates of the piano intro of “For Elise”. Here, the “baseline” changes more over the time and has an irregular waveform (better shown in Fig.3.2b). The

smaller amplitudes represent a much lower variance of the zero-crossing rates, what is characteristic for the nature of music. The baseline of this piano-piece of music is very similar to the piano-sequence shown in [34], and the one high peak in Fig.3.2 at the left corner is caused by the only unrhythmic free-played piano intro in “For Elise”, which demonstrates again the difference of the zero-crossings for periodic and non-periodic patterns in a sound file (after the piano intro, the periodic music structure can be identified by the small range of the amplitudes). This example explains also the high peaks in electro- or techno music (Fig.3.2e and Fig.3.2f): while the basic beats in such files are nearly periodic (low values of the variance and small range of the amplitudes), some overlayed rhythm-elements (or sound effects) will cause the peaks. This effect can also be heard in the sound samples.

Fig.3.2d shows the zero-crossing rates of the “james bond” movie theme with the zcr-amplitude peaks caused by the own rhythm of the lead guitar, which is different and independent played to the periodic basis rhythm of the other instruments. Fig.3.2g shows the zero-crossing rates for speech only: there are only peaks as mentioned before. Those peaks are two times higher as in all cases of music (many peaks up to 0.7), and so there is a significant threshold value, that enables separating speech from music. Further, Fig.3.2.g shows the zero-crossing rates of a news speaker, the only case, where speech can not be classified as speech by the zero-crossing rate estimator. We see the peaks in the range of music and the music-alike baseline. This “speech combined with background music” pattern is caused by the very periodic speech pattern (pronouncing and speech velocity) of professional news speakers and can be heard in the audio-files. In each news-audio file from the test-setup, this speech- and zero-crossing rates pattern appeared. This effect of mixed-audio (speech with music) can also be seen in [34], where the variation of average zero-crossing rate with the percentage of speech in audio is demonstrated. Further, as can be seen in the following Fig.3.2h., if any background music or sound effect appears in combination with speech (non-speech case), the peaks will fall back to the amplitude range of music and are classified as non-speech content.

The following figures show typical zero-crossing rates (“ZCR-Amplitudes”) for each frame of audio signals with different contents. The audio files were chosen to demonstrate the effect how the zero-crossing rate estimator classifies.

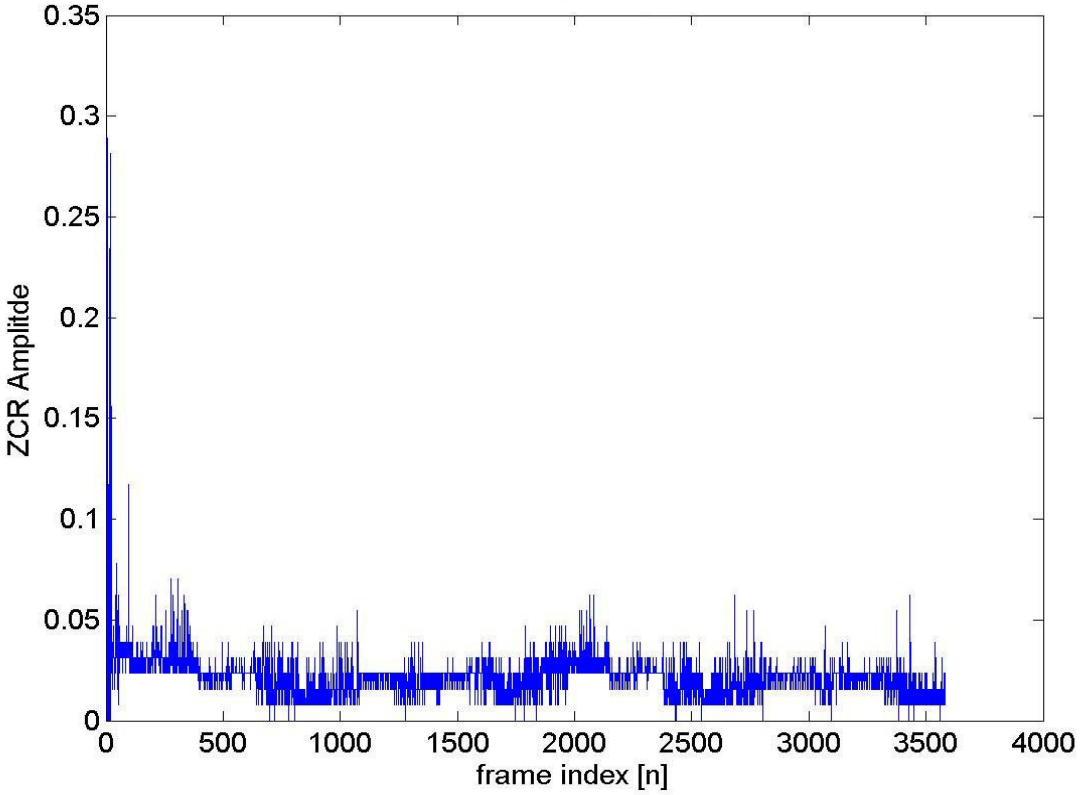


Figure 3.2.a: Zero crossing rate in each frame of a classic music file

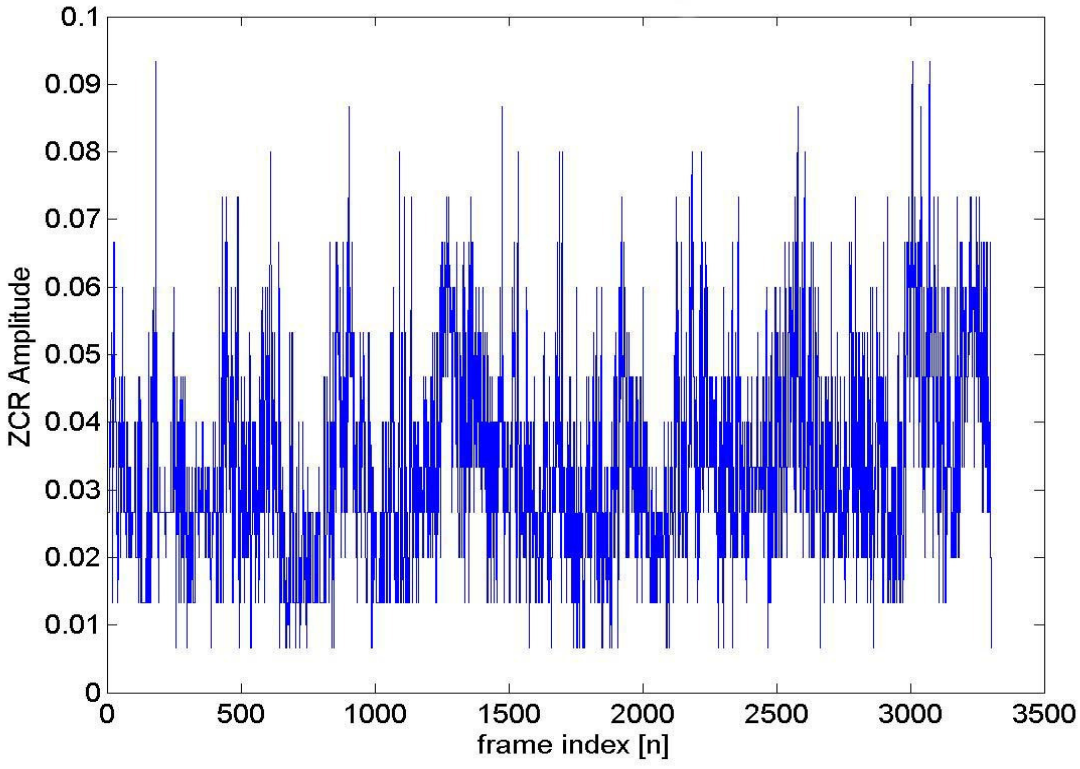


Figure 3.2.b: Zero crossing rate in each frame of an orchestra music file

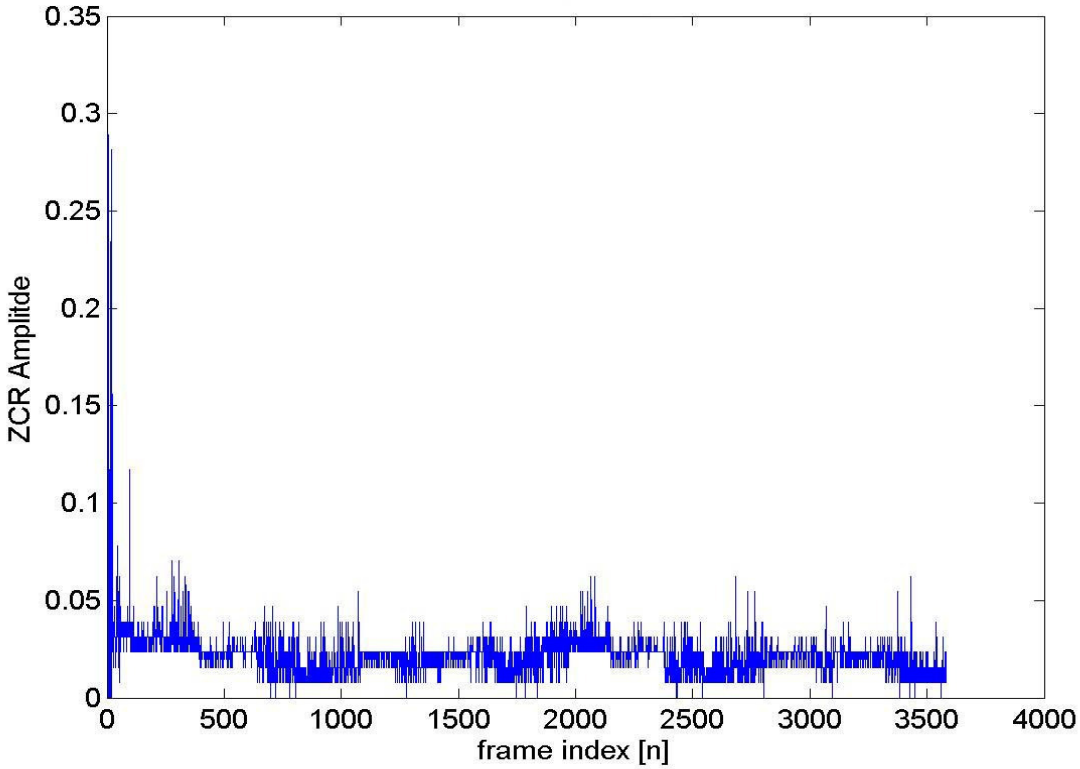


Figure 3.2.c: Zero crossing rate in each frame of a pop music file

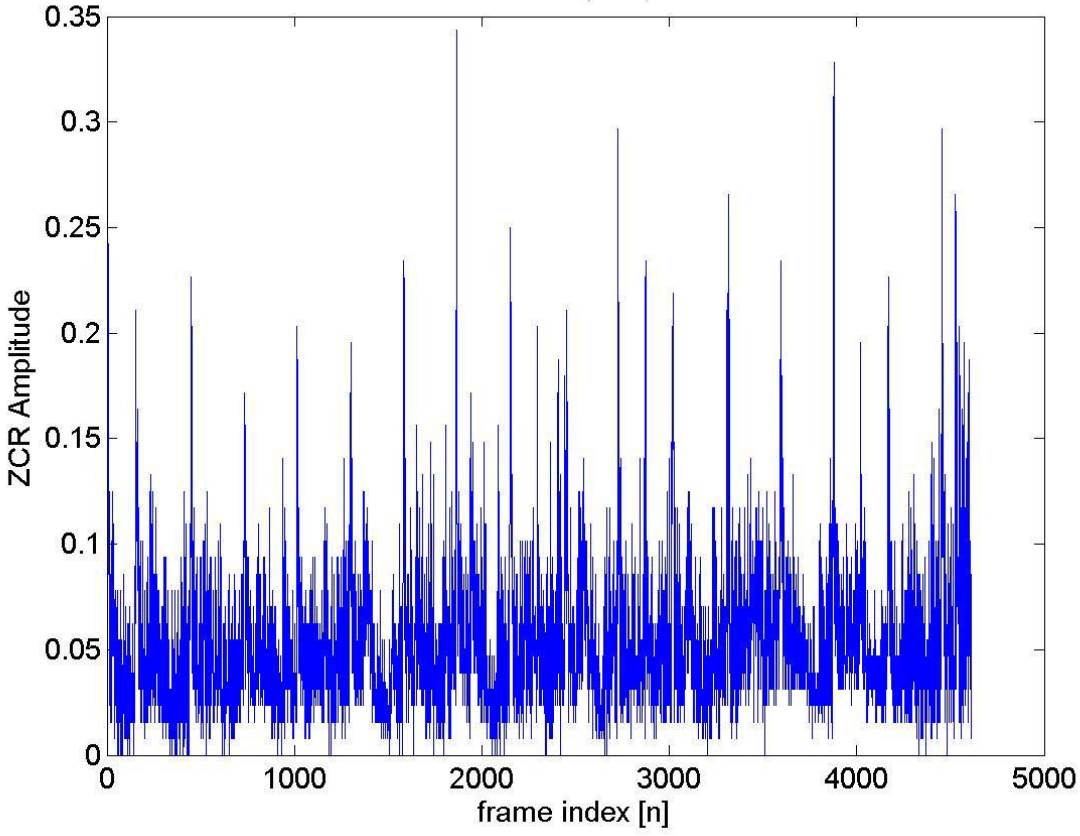


Figure 3.2.d: Zero crossing rate in each frame of a rock music file

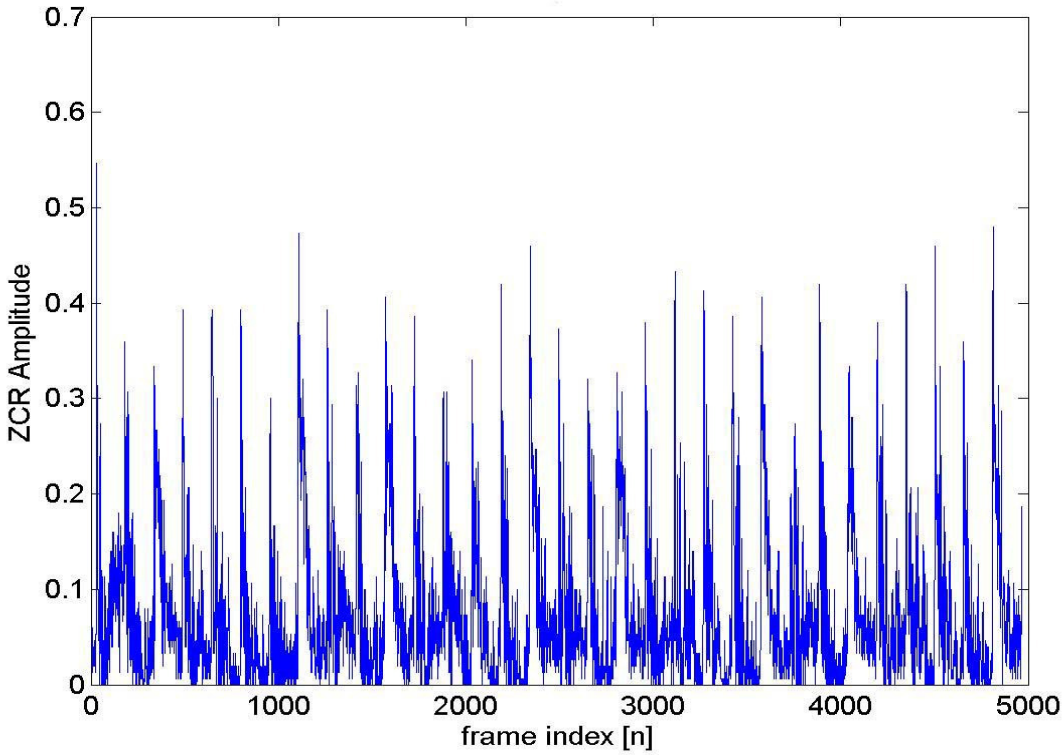


Figure 3.2.e: Zero crossing rate in each frame of a techno music file

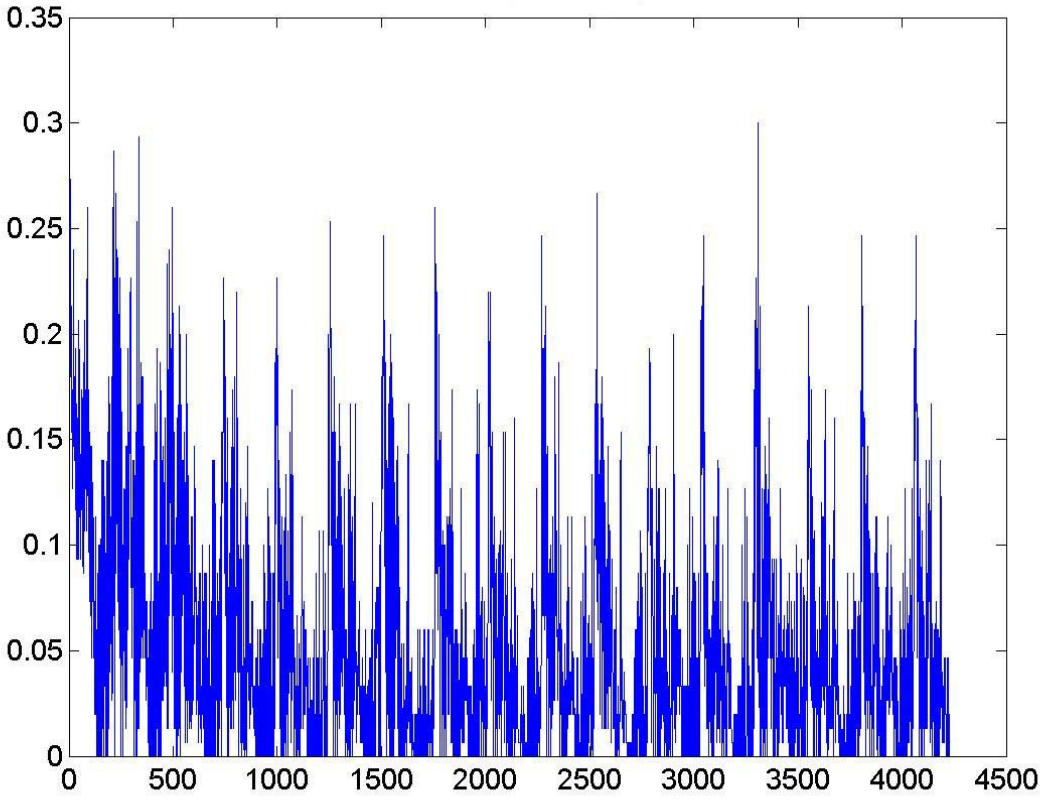


Figure 3.2.f: Zero crossing rate in each frame of an electro music file

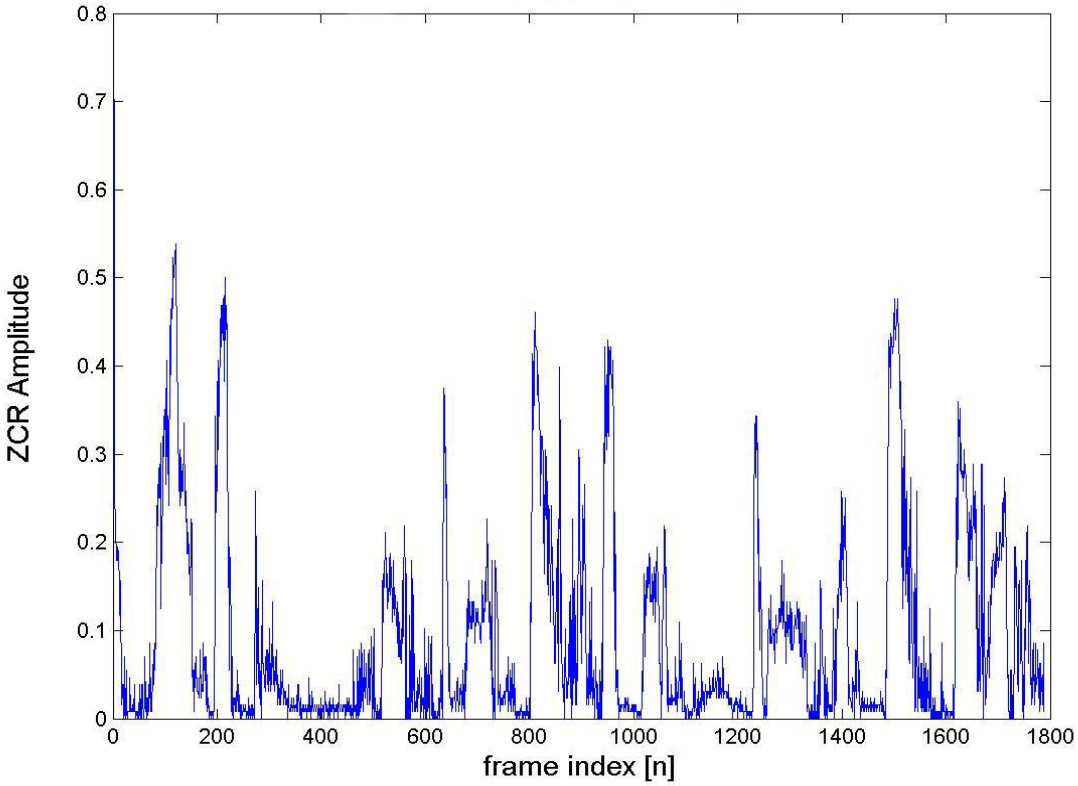


Figure 3.2.g: Zero crossing rate in each frame of a speech file

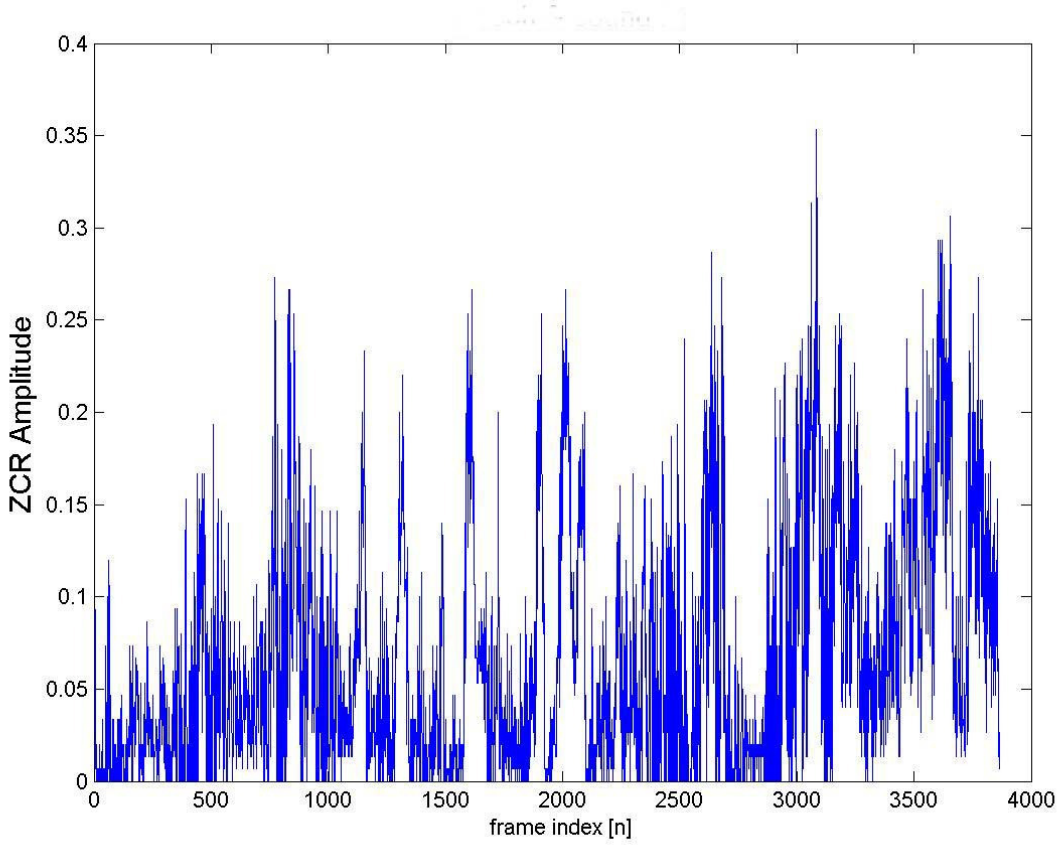


Figure 3.2.h: Zero crossing rate in each frame of a speech file with background sound fx

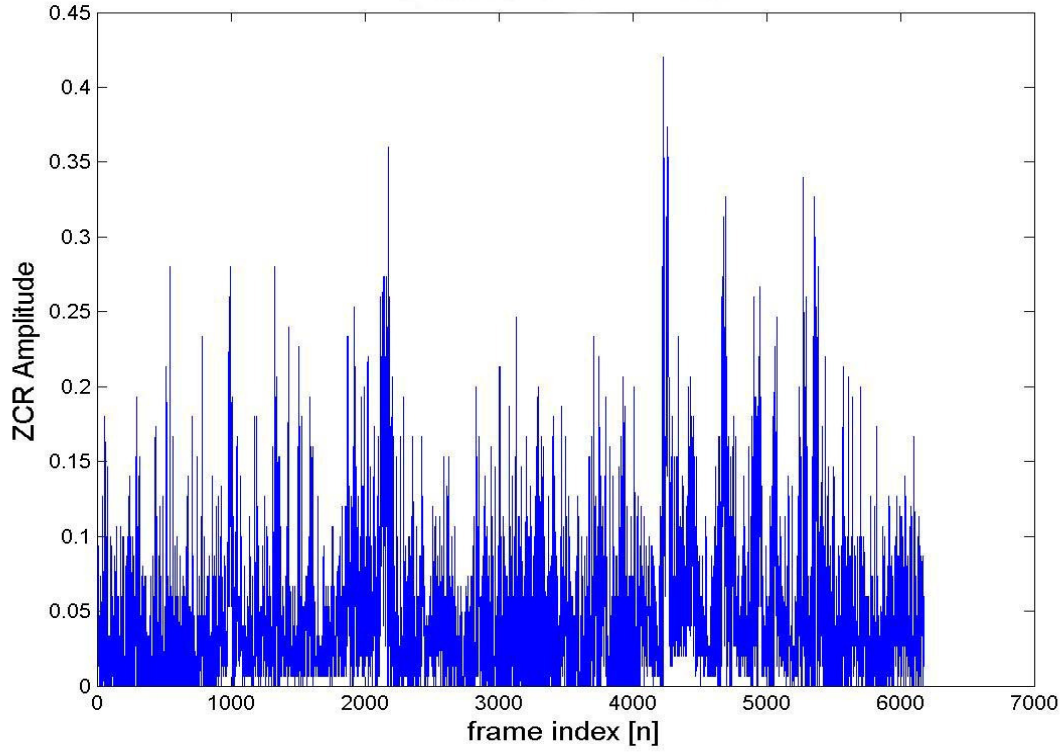


Figure 3.2.i: Zero crossing rate in each frame of a speech file with background techno music

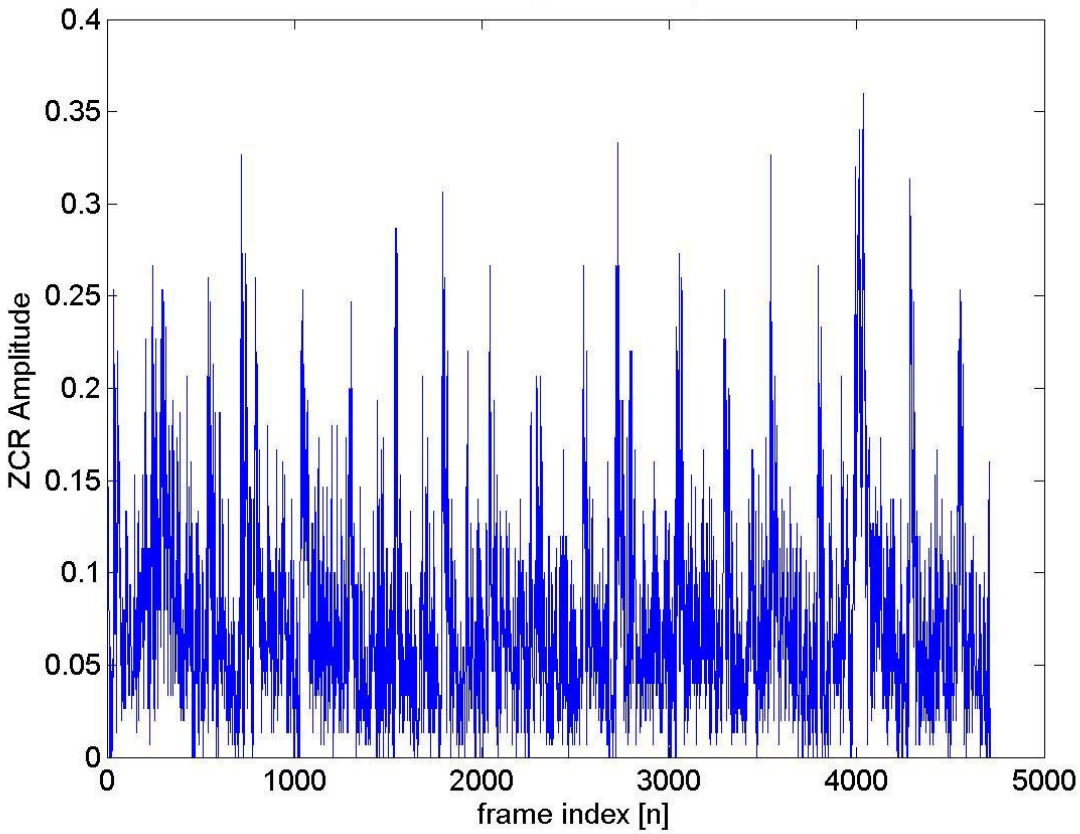


Figure 3.2.j: Zero crossing rate in each frame of a dance music file with female voice

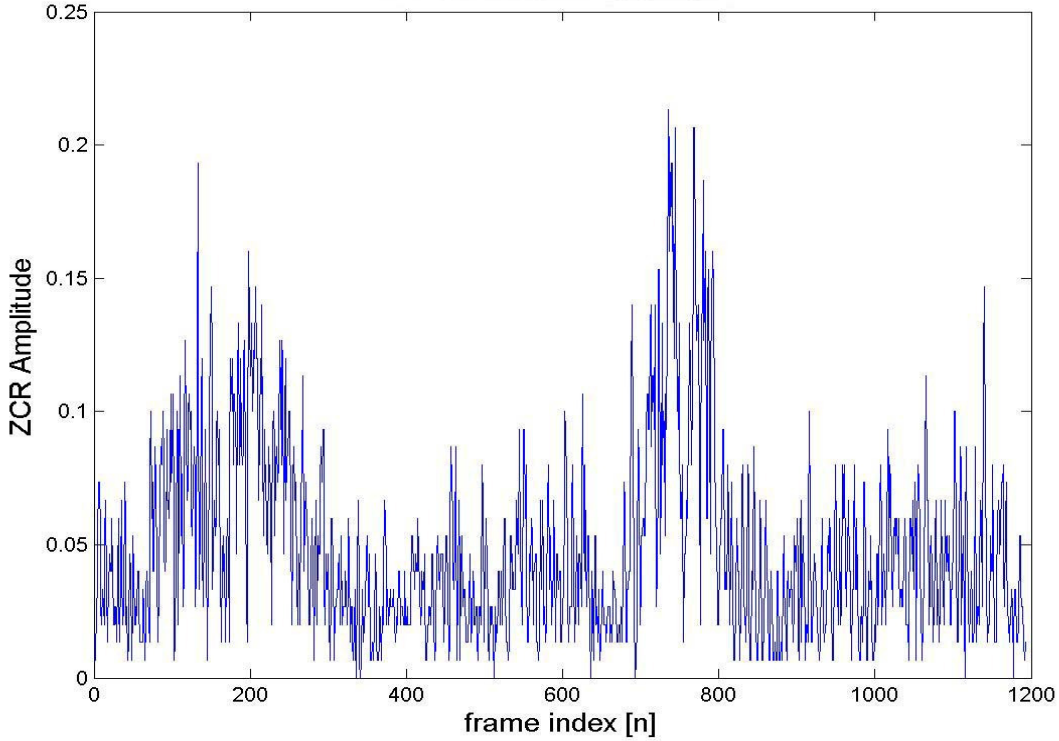


Figure 3.2.k: Zero crossing rate in each frame of a speech file with orchestra background music

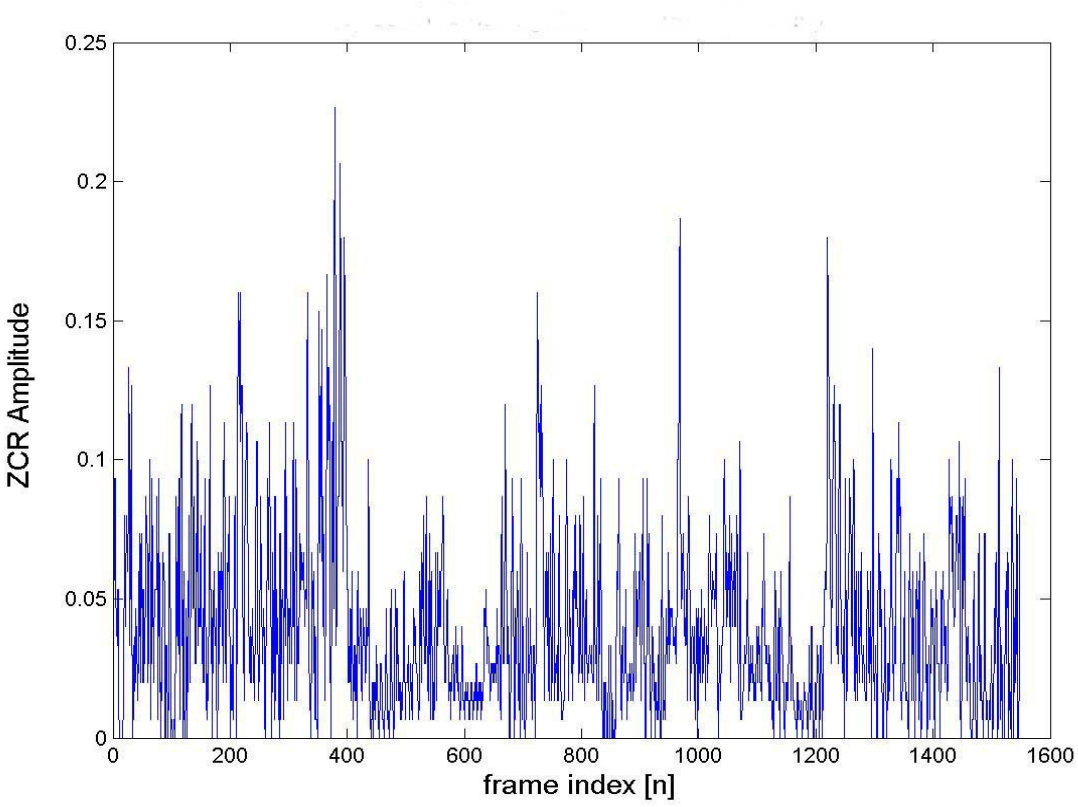


Figure 3.2.l: Zero crossing rate in each frame of a speech file with background classic music

3.2.3 Music test results

- The music test setup contains 182 audio files of different lengths (100ms … 7s), audio quality and different music content (classic, orchestra, pop, rock, ethno, spheric, dance, techno and sound fx's). Only five of them were identified as speech. -> 97.2527 % are identified as music.

3.2.4 Speech test results

- The speech test setup contains 165 speech files of different lengths (500ms … 7s), quality and content (speech only, speech with background music / fx's, speech of news moderators and speech from professional studio speakers)

- Speech only, professional studio speakers: 75% identified as speech. An extended test-setup with more different professional speakers will be investigated to increase this value

- Speech with background fx sounds taken from video source: 100% identified as music

- Speech with background music (all styles) taken from video source: 97,44% identified as music

- Speech only from video source: totally 70, 27% identified as speech (36 files), but here it depends on the velocity of the speaker's voice: news speakers speed are very fast and has a very music-likely "rhythm", as can be heard in the sound-samples: 100% are identified as music. This "music behaviour" of the zero-crossing rates is also represented in the zero-crossing graphics. In the case of normal speakers (speech only, video-source): 96% are identified as speech

3.2.5 Conclusion: zero-crossing rate estimator

An estimator based on zero crossing rate identifies music and speech with background music and effect sounds as music, which is ideal for analyzing cinema trailers, advertisements, documentations, or, in the simplest case, the introduction of a music band by a moderator (speech content -> music content -> music content with voice).

One great advantage of a zero-crossing estimator is the short identification time (correct identifications for 500ms speech segments and 100ms music segments as shown in the tests). Further, the calculation of the zero crossing rate is very simple and fast, it uses characteristics of the original signal in the time domain (-> no further transformation).

In relation to audio codecs and audiovisual quality, an estimator based on zero crossing rate enables to choose automatically the right coefficients for the parameter of the audio codec and audio content specific audio quality metric parameter. For example, in case of AMR WB and AMR NB coded speech content, audio codec and audio content specific coefficients for the model audio quality output parameter auditory distance AD can be chosen from a table, or for the case of music and music with speech coefficients, corresponding to AAC, coefficients for the integrated frequency distance parameter IFD [1] and the disturbance indicators D_ind and A_ind. The disadvantage of such an estimator is the identification of news-speakers (speech only), while the voice of news speaker has a rhythmic similar to music. For coded audio content, as described in chapter 5, the same method for audio content classification works also for news speaker, based on a modified version of the coded audio file and a variation of the threshold value.

Analyzing several news-clips, there can be said, that most of them follow the same structure: music only as intro -> music with speech for the introduction -> then, most of the time only speech, interrupted by sequences with speech and background noise / fx's. Music with speech sequences during a news-scenario are rare. Based on this average structure of news, further test should prove, if such clips need audio content classification or if the information about the video content [27] (news -> most time speech) is enough to classify them as speech, see chapter 6. On the other hand, video content classification, based on audio scene characteristics, is possible, as described in chapter 6. Another method is based on the usage of subband

energy estimation, in which the energy of a given frequency interval in the frequency domain is calculated and compared to a threshold, dividing speech from non speech content. In the case of news, subband energy estimation leads to better classification results than in the case of non speech content. Test results of the zero crossing rate audio content classification are given in appendix D.

3.3 Audio content classification based on subband energy estimator

3.3.1 General aspects

Content classification based on subband energy estimation uses sound characteristics in the frequency domain, as presented, e.g., in [20], [22], [23], (2,5), while [24], [25] gives a good overview of frequency-domain features. So the computation for this estimator is more complex and not so easy as for the zero crossing rate estimator, if the estimator works alone for itself. Otherwise, the implementation of this algorithm in a program which already works with signal-framing, windowing, FFT and frequency splitting (normally most of programs which deals with digital audio signal processing) is very easy and the algorithm will be reduced only to calculate the subband energy ratio [24] comparing to the threshold level, defined by the tests-results. Similar to the zero crossing estimator, an audio file is divided into frames of 100 samples with 50% overlapping, windowed with a hamming window and each frame is transformed to the frequency domain. The whole frequency spectra is divided into 4 subband intervals, similar to [24], given by the half of the sampling frequency (22050Hz). For each frame the subband energy ratio is calculated. The final chosen subband intervals follows the bounded intervals in [24], and are given by:

- Subband 1: (0-2756) Hz

- Subband 2: (2756-5512) Hz

- Subband 3: (5512-11025) Hz

- Subband 4: (11025-22050) Hz

Variations of the subband boundes were made (e.g., to concentrate them on the frequency range of the speech formants), but they offered no better classification results or threshold bounds, comparing the subband energy ratio of each subband to find a threshold for this ratio to separate speech from music. For this, the subband energy ratio is calculated by dividing each subband energy to the total energy of the spectra. As mentioned above, every single frame of the audio signal is transformed to the frequency domain by Fourier and represents further the distribution of the spectral energy over this audio frame. By this way, a comparison of the spectral energy distribution of each audio frame is possible, which shows, how the spectral energy will change in time during the whole audio file. In case of music, periodic pattern will appear. A much more complexer method to find characteristic periodic patterns in audio content is demonstrated in [26]. In this method, the envelope of an audio signal, which can be seen as an amplitude-modulated wave, is extracted and the frequency spectrum of this "modulation"-wave is calculated. The investigation of statistic information (mean or standard deviation of the spectral energy distribution from frame to frame) to design an estimator to detect periodic and rhythmic structures as the main difference of music and speech was also done, but gave no further significant threshold values for classification.

The test setups for subband energy estimator are the same as for the zero crossing rate estimator, which enables a comparison of them. An useful threshold value, similar to equations (3) and (4) in [28], for the subband energy ratio was found at 0.93 to separate speech from music:

$$SER = \frac{\int_{\omega_1}^{\omega_2} P(\omega)d\omega}{\int_{0}^{\omega_{max}} P(\omega)d\omega} \qquad (3.2)$$

where

SER …… subband energy ratio

ω_1, ω_2 …. specific subband interval bounds

ω_{max} …… highest frequency of the spectrum

$P(\omega)$ …… Power at frequency ω, where $P(\omega) = |F(\omega)|^2$

$F(\omega)$ …… FFT coefficients

3.3.2 Audio content classification results for speech and non speech content

The same kind of music content was used: classic, orchestra, pop, rock, ethno, spheric, dance, techno and sound fx's.

- CD, classic: 100% identified as music
- CD, Video, rock: 60% identified as music
- CD, Video, ethno, wave, spheric: 45.45%
- CD, Video, sound fx's: 42.857% identified as music
- CD, Video, orchestra: 40.47% identified as music
- CD, Video, pop : 14.29% identified as music
- CD, Video, beat: 27.27% identified as music
- CD, Video, techno, dance, electro: 0% identified as music

The dependence of the music style of this estimator shows, that more rhythmic elements in a sound destroys more the periodic pattern of the spectral energy distribution. This can be the reason why the spectral energy distribution of such music files are very similar to those of speech and reduces the quality of such an estimator to separate speech from all kind of music styles.

3.3.3 Speech test results

The speech test setup is the same as for the zero crossing estimator. Based on the spectral energy distribution of speech as described before, the estimator will identify an audio content as speech in every case of speech only, speech with background music or fx's. Only 9% of all 165 test files of speech only, speech with background music and fx's were identified as music. For this reason such an estimator could be used alternatively for uncoded news. Detail results for audio content classification, based on subband energy ratio in comparison to zero crossing rate based audio content classification are given in appendix D .

3.3.4 Conclusion: subband energy ratio estimator

With subband energy estimation, it is not possible to identify speech with music as music (which is relevant for audio codecs and audiovisual quality metrics) and the only audio sub content category, which can be separated from speech, is classic music. Another disadvantage of this estimator is the instability, that means, that, based on the algorithm, subband energy ratio of some audio files could not have been calculated (3.48% of music and 3.363% of speech test setup).

3.3.5 Final conclusion: audio content estimators

Finally it can be said, that for all kind of uncoded audio content (speech only, speech with background music and fx's, and songs) except for news-speakers the zero crossing estimator is a suitable classifier for audio content classification: it identifies all different styles of music and also the combinations speech and music (which is important for AAC codecs) as good as all non-news-speakers. Further, the zero crossing rate estimator enables automatic content classification for an audiovisual metric by analyzing very short audio file segments, and this estimator is easy to implement. Subband energy ratio estimator will be suitable for news speakers, but not for separating music from speech and so this estimator is not preferable for the design of an automatic audiovisual metric. This estimator is easy to implement in most of digital audio signal processing programs.

3.4 Audio content classification for video sequences

In relation to audiovisual quality for multimedia content it is necessary to estimate the audio content in one or more video sequences. Therefore, the zero-crossing estimator will classify the audio content as speech or non speech. The main difference to audio only content classification is the changing of the content between several video cuts (video sequences). The estimation time intervals for audio content classification here are much shorter in which

the audio content can be analyzed and the estimation classification result depends on every single video sequence content.

3.4.1 Audio content classification based on video cut time points

To estimate the audio content in a video sequence it is necessary to find the start- and end points of every video scene in a video file. By using a scene change detection tool, it is possible to transform the frame number of the video cuts into time points in seconds to find the equivalent scene change time points in the audio track to synchronize the multimedia components audio and video. After that, there are three methods for the estimation time interval length in each video sequence, depending on the video content:

- each single sequence will be analyzed during its whole length

- each single sequence will be analyzed during 30% and 50% of its length

- the shortest cut time difference or the average sequence length of all sequences

3.4.2 Test results: music videos

All of the music videos from the setup are classified for 100% as music (without lead in / lead out effects). Appendix contains the table with the test results.

3.4.3 Test results: music documentary

Music documentaries enables the simplest case to investigate video-cut based audio content estimation:

- First sequence: speech (introduction of the artist)

- Second sequence: music (song)

The test files “roy black” and “angel_end”, which follow this structure, were classified 100% as music. Test files “angel_start” and “come_undone” were classified as 90% and 89% as music. Those results based on the fact, that in this test files the music of music sequences fades in and out into the speech sequences. Appendix E contains the table with the test results.

3.4.4 Test results: cinema trailers

The test-setup for cinema trailers consists of:

- Cinema trailers with non-speech content (music or speech with background music in every scene) only, fast and slow scenes

- Cinema trailers with speech and non-speech content (music or speech with background music in every scene), fast and slow scenes

For cinema trailers, lead in and lead out effects at the beginning and the end of the whole trailer causes false classifications for the first and last video sequences and were not used in the analyzation process.

The tests of audio content classification for weather and advertisements are done for coded audio content in chapter 6. Appendix E contains the table with the test results.

Chapter 4

Reference based and reference free audio quality estimation

4.1 Reference based audio quality estimation for different coded audio contents

Previous works [1-5] and the results from the subjective MOS listener tests (cf. 2.2.2.) show, that for mobile streaming services AMR WB / AMR NB are the most suitable audio codecs for speech content and AAC for music content. Further, it is possible to reach the same perceived audio quality for a coded speech content file using AMR WB with codec settings 15.85kbps, 16kHz instead of the advanced audio codec AAC, working at 24kbps with sampling frequency 22.05kHz. Together with the characteristics of each audio codec, zero-crossing rate estimation allows the identification of the content of an encoded audio file.
Further, in relation to audio quality metrics and perceptual models [1-11], it is possible to design one automatic audio quality metric with different metric parameter coefficients for each encoded content type and specific objective quality measurement system model output parameter to estimate the audio quality of a coded audio file. Therefore, audio codec and audio content classifications are necessary. Once, the audio codec and audio content are estimated, the specific audio quality metric coefficients for the specific objective quality measurement model output parameters can be chosen from a table [1-3]. While the coded audio content can be classified by zero crossing rate estimation, the audio codecs can be classified by their individual setting characteristics or, as for the case of AAC and AMR, by comparing the different lengths of the original and encoded audio file (reference based audio codec classification):

- AMR decreases the length of the coded signal: length(orig) - length(degr) > 0

- AAC increases the length of the coded signal: length(orig) - length(degr) < 0

Based on the information about the audio content, determed by audio content classification, audio codec, and original audio file as reference, one audio quality metric with different coefficients for each audio codec and audio content can be designed (see chapter 2).

4.2 Reference free audio quality estimation for different coded audio contents

The disadvantage of reference based audio quality estimation methods in relation to complexity and computation power of the quality assessment methods is the usage of original, uncoded or coded reference files, consisting of different audio contents. While all of those reference depended algorithms are finally based on the comparison between an original, uncoded or coded, degraded audio file as reference information, a calculation of the perceived audio quality without information about such reference files is not possible. Moreover, a collection of reference files with typical content or codec characteristics stored for comparison will increase calculation time and the complexity of the audio quality metrics and systems. Such reference based audio quality classification systems are always limited by the available reference file collection and their classification characteristics. All of those disadvantages can be avoided by the development of reference-free audio quality estimation metrics or systems. Furthermore, reference-free metric parameter, which are determed directly from the frequency spectrum of an unknown coded audio file without the usage of objective quality measurement model outputs or algorithms, will decreases the complexity of the whole estimation / calculation / classification process. This reduction is possible, if all design processes of perceptual audio quality models and metrics are based on the statistic results of subjective listener tests (MOS scale values, see sub chapter 2.2.1). So, the reduction of an audio quality metric, consisting of parameter extracted by objective quality measurement systems and their coefficients, to a single audio quality equation, consisting of audio quality parameter extracted from the degraded, coded audio file frequency spectrum without specific reference information will lead to the ideal case of lowest “metric” complexity, computational power and calculation time, mathematically formulated as AQ = aqpc * aqfp, where aqfp is the extracted audio quality feature parameter, aqpc the audio quality parameter coefficient, and AQ the resulting audio quality. For this case, the audio quality equation parameter coefficient aqpc must stand in relation to the mean result of how listeners would classify the perceived audio quality of test files consisting of the same audio characteristics. Once, one or

more parameter coefficients are found, such a relation is possible by a simple mapping function between the audio quality parameter coefficient aqpc and the mean of the subjective MOS test results for a specific coded audio content type (speech or non speech content).

As Fig.2.1 shows, parameter extracted by objective quality measurement models are always based on reference values or information about the original, uncoded file, e.g., in case of PESQ, the auditory distance AD between the original and degraded audio file or the integrated frequency distance IFD between the original and degraded audio file. In case of reference free parameter extraction, the extracted parameter is deviated directly from the frequency spectrum of the coded audio file without further reference information about the original, uncoded file or audio codec settings. The results of the subjective MOS audio listener tests show the influence of each single audio codec with its individual performance settings bitrate and sampling frequency on the perceived audio quality of different coded speech or non speech content. While in most of the reference based audio quality classification systems the parameter for the audio quality metric are extracted by objective quality measurement models, reference free audio quality equation parameters are extracted directly from the coded audio file without reference information, and their coefficients are determed directly by the results of subjective MOS tests. So, the quality of the classification process is given by the quality of the MOS test setup (number of different coded audio content, number of test listeners, ...) and its results. Once a MOS test session is done, an individual reference free audio quality feature parameter coefficient for that specific test setup can be extracted by using specific time- or frequency domain algorithms, representing all characteristics which influences the audio quality of each single test file.

Table 4.1 resumes the different coded audio test files from the subjective MOS listener tests and their individual settings, representing the main characteristics of the chosen MOS test setup to find a suitable audio quality classification metric for different coded audio content:

audio codec	bitrate [kbit/s]	sampling frequency [kHz]	speech stadt.wav		classic music haydn.wav		other music FA.wav		fx sound stadion.wav	
			mean (MOS)	MOS	mean (MOS)	MOS	mean (MOS)	MOS	mean (MOS)	MOS
AAC	8	8000	1.14	1	1.66	2	1.1	1	1.476	1
AAC	16	16000	3.2	3	3.76	4	2.9	3	3.66	4
AAC	20	16000	4.38	4	4.66	5	4.38	4	4.43	4
AAC	24	22050	4.85	5	4.95	5	4.85	5	4.66	5
AMR WB	6.6	16000	1.43	1	1	1	1.095	1	1.1	1
AMR WB	8.85	16000	2.23	2	1.1	1	1.38	1	1.476	1
AMR WB	12.65	16000	4.33	4	2.66	3	2.85	3	3.95	4
AMR WB	15.85	16000	4.81	5	3.47	4	3.2	3	4.52	5
AMR NB	4.75	8000	1.14	1	-	(1)	-	(1)	-	(1)
AMR NB	7.97	8000	1.91	2	-	(1)	-	(1)	-	(1)
AMR NB	12.2	8000	2.71	3	-	(1)	-	(1)	-	(1)

Table 4.1: Subjective MOS listener test setup and results.

mean(MOS) ... mean value of the rated MOS from subjective listener tests for each audio codec and audio content

MOS ..... mapped to valid MOS scale value

While only AMR NB coded speech content were included in the MOS test setup, all AMR NB coded non speech content were classified as “1 ... BAD”, based on its sampling frequency 8kHz in comparison to AAC coded non speech content with sampling frequency 8kHz and 8kbit/s. All .3gp files were further encoded to the .wav standard with sampling frequency 44.1 kHz to avoid further audio quality distortions. The final bitrate of those .wav files are increased or converted to 256kbit/s. This finaly bitrate value of 256kbit/s causes no further audio quality degration during the bitrate conversation process. The loss of the original bitrate information in the encoded .wav file leads to a bitrate reconstruction mechanism to reconstruct the original audio codec bitrate from the 256kbit/s coded .wav file version, as

described in sub chapter 5.2.3. This bitrate reconstruction mechanism is essential for the automatic reference free audio quality classification in that way, that the individual chosen bitrate has one of the strongest influence on the perceived audio quality, more than the sampling frequency, as shown in Table 4.1. For example, the perceived audio quality of AAC coded speech content with 20kbit/s is better rated than the quality of AAC coded speech content with 16kbit/s, using the same sampling frequency of 16kHz.

Such a reference free audio quality classification design, based on a MOS test setup with individual chosen audio content, audio codecs, different bitrates and sampling frequencies is flexible for all other kind of other individual chosen MOS test setups, consisting of MOS test setup characteristics audio codecs, bitrates, sampling frequencies and content types. That means, that this reference free design method or strategy can be extended, focused or applied on every other individual chosen test setup (universal reference free design method). There is no difference in the design strategy, if the MOS test setup consists of other individual chosen audio codecs, bitrates, sampling frequencies or audio content, while the design method is based on the information extracted from the specific chosen MOS test setup and on no other kind of other reference information.

Chapter 5

Reference free audio quality estimation system

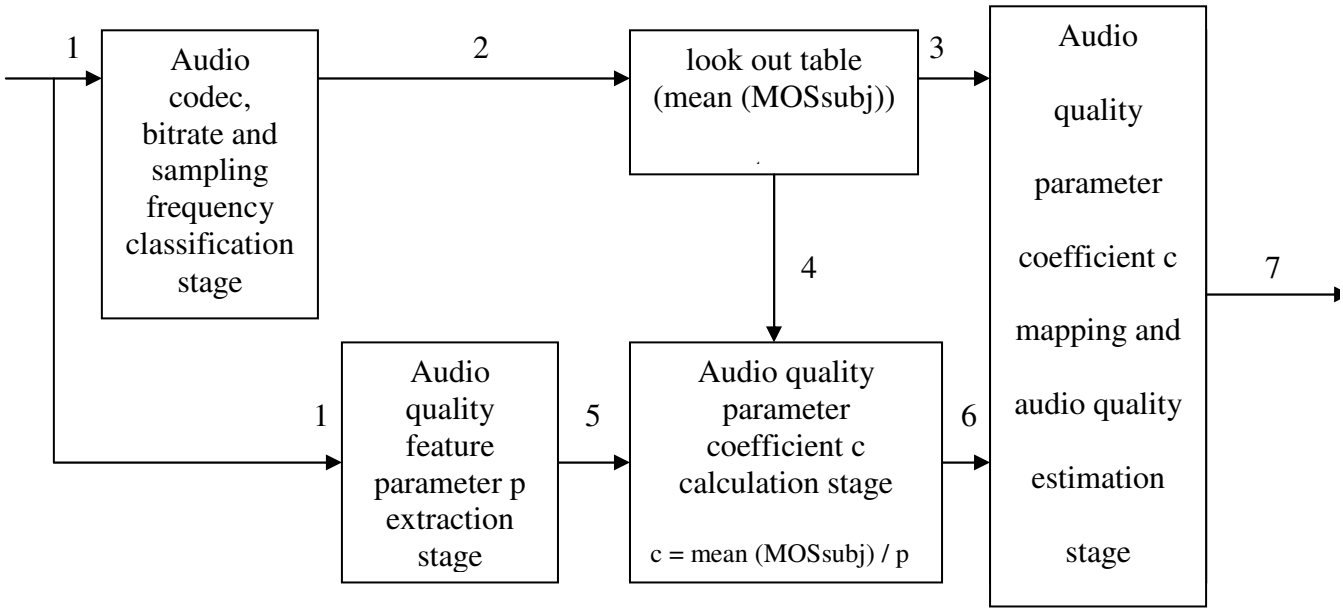
5.1 Overview of a reference free audio quality estimation system

In reference free audio quality estimation systems, the audio quality of a coded audio file is predicted by the following units:

- audio codec classification stage:
  - audio codec classification stage to identify the unknown audio codec of the audio file
  - audio codec bitrate recovery and sampling frequency classification stage
- audio content classification stage for coded audio content
- audio quality parameter feature extraction stage for different kind of coded audio content and audio quality parameter coefficient prediction
- audio quality parameter coefficient to MOS scale value mapping stage

Each of those stages extracts specific feature parameter from the coded audio file for further classification processes, based on threshold comparison, to enable reference free automatic audio quality estimation metrics. For the case of reference free audio quality classification, those classification processes are also necessary for mapping the audio quality parameter coefficient c to its corresponding MOS_{Apred} value. In the first two steps of such an audio quality classification system, the unknown audio codec, bitrate and sampling frequency are identified within the audio codec characteristic classification unit, before the audio content can be classified. After that, an audio quality parameter is extracted by the feature extraction

stage and its audio quality parameter coefficient is calculated for each kind of coded audio type. The mapping of the predicted audio quality parameter coefficient to the equivalent MOS scale value is done in the last step by mapping the audio quality parameter coefficient c to a valid MOS scale value mapping unit. Fig.5.1 shows the block diagram with all of this main function blocks, which may differ from system to system:



1 … unknown coded, raw audio file (.wav)
2 ... audio codec, bitrate, sampling frequency, and audio content identifiers
3 ... round(mean(MOS_{subj}))
4 ... mean(MOS_{subj})
5 ... audio quality parameter p
6 ... audio quality parameter coefficient c
7 ... predicted perceived audio quality (MOS_{Apred} = round (c * p))

Figure 5.1: Reference free audio quality estimation system.

The first stage classifies the audio codec, the audio codec bitrate and the sampling frequency of the unknown coded audio file. With those characteristics it is possible to choose the mean value of the audio codec specific subjective MOS, corresponding to the specific audio codec and its characteristics, using a lookout table. From the unknown coded audio file, an audio quality feature parameter p is extracted at the audio quality feature parameter stage. Predicting

the MOS_A using a linear equation, an equation coefficient c is calculated as the ratio of the chosen mean value of the subjective MOS and the audio quality feature parameter p.

Finally, this audio quality parameter coefficient c is mapped to a valid integer value in the specific, audio codec characteristic range at the audio quality estimation stage. The correlation between the rounded version of the subjective MOS mean value and the so predicted MOS_{Apred} can be calculated by the Pearson linear correlation factor or by the characteristics of a correlation vector. Fig.5.2 shows the same audio quality estimation system in a more detailed version, while all function blocks from Fig. 5.1 are described in the following chapters:

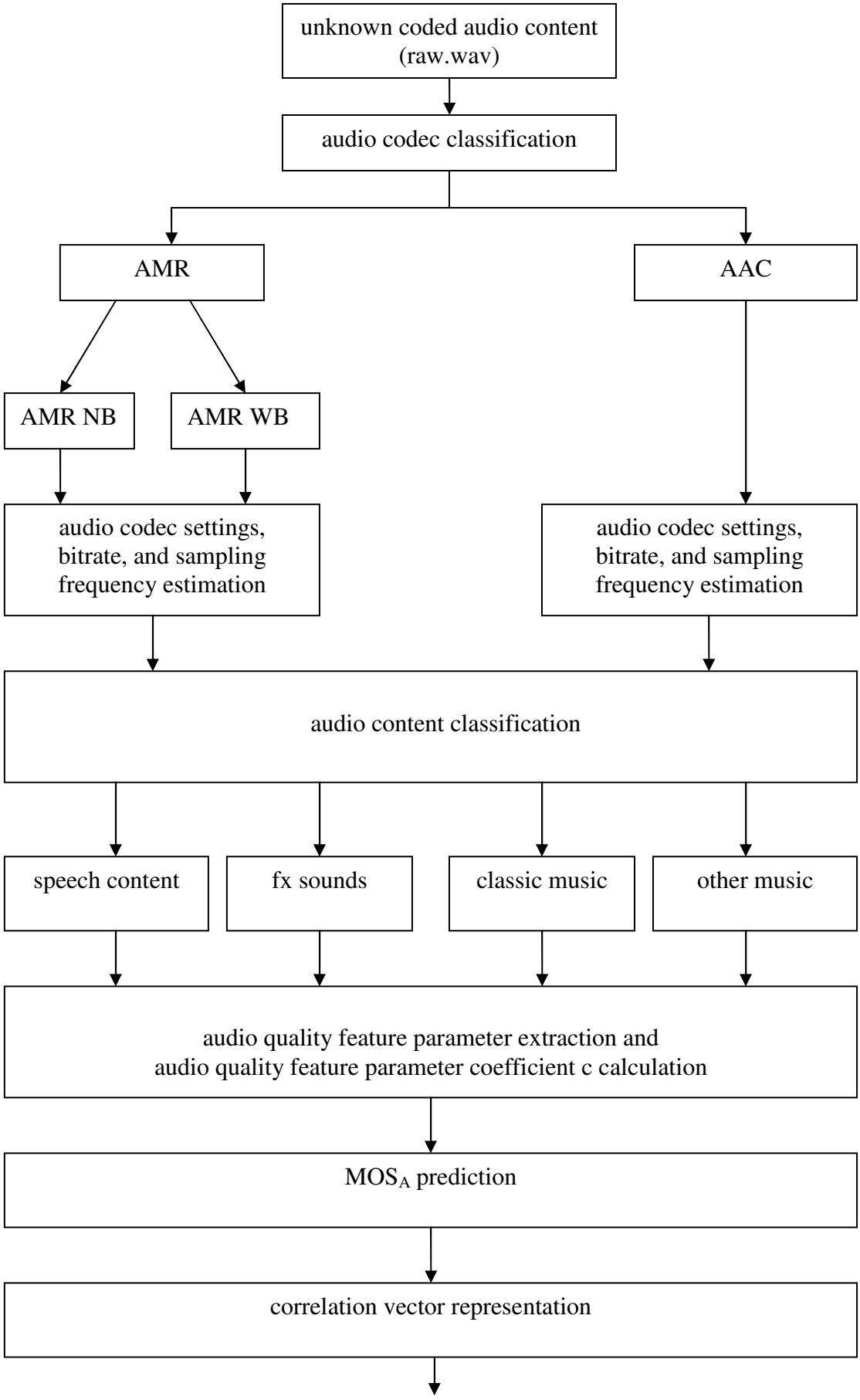


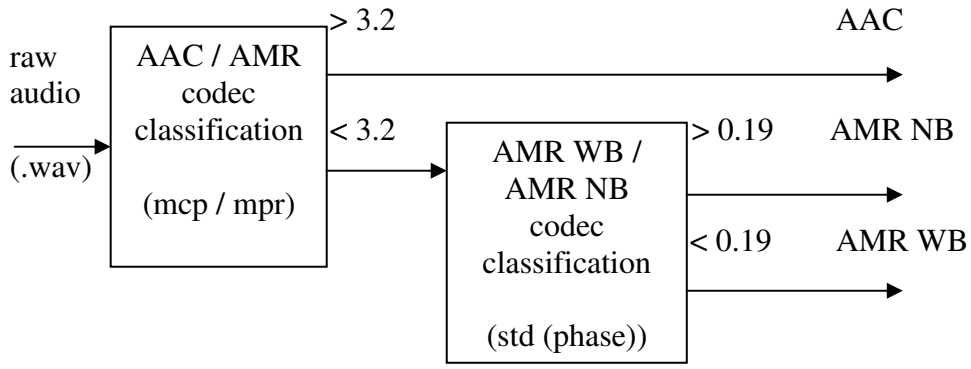
Figure 5.2: Flow chart of reference free audio quality estimation system.

5.2 Reference free audio codec characteristic classification stage

Each reference free audio codec classification system tries to find specific audio codec information or characteristics in the time- or frequency domain of the unknown coded audio content. Those audio codec characteristics are caused by the different audio codec performance settings, e.g., the audio codec bitrate and sampling frequency. Furthermore, both of them have the strongest influence on the perceived audio quality of a coded audio file, as can be seen in the results of the subjective MOS listener test, or in Fig.5.19. While non reference free audio codec classification systems identifies the unknown audio codec using stored reference information about each audio codec and audio coded content type, reference free audio codec classification systems are identifying the audio codecs by analyzing just the unknown coded audio file without the knowledge about the original, uncoded audio file.

Reference free audio codec classification systems use information extracted from the time- or frequency domain of the unknown coded audio file. While there can be extracted several different information parameter from this domain (extracted feature parameter), classification tests with those parameter decreases the number of optimal classification parameter for an audio codec classification system. Examples of such extracted feature parameter from the frequency domain are the mean centre frequency, the mean bandwidth, their ratio, the mean centre phase, or the mean phase range. Feature parameter examples extracted from the time domain are the standard deviation and mean value of the zero crossing rate. The extracted feature parameters (classifiers) were investigated, if and how they reflect specific audio codec characteristics in a coded audio file to create a relation between the classifier and the specific audio codec. For example, there are different specific audio codec characteristic classifiers or identifiers for the classification group AAC and AMR and the sub classification group AMR WB and AMR NB. Such an audio codec classification system, based on extracted feature parameter from the unknown coded audio file, is reference free and audio content independent, that means, the classification parameter stands in no relation to the coded audio content. Using such extracted feature parameter, the audio codec of unknown coded audio content can be identified. Furthermore, extracted feature parameter can be combined to classification vectors for classifying other audio codec characteristics as bitrate or sampling frequency, which reduces the number of classification parameter for a reference free audio codec classification system. One specific classifier can be seen as an optimal classifier within the

whole classification system, if the classifier can be used for more than one classification process. Such an optimal classifier reduces the complexity of the whole audio codec and audio content classification system. The expressions for the classifier are given in equations (5.1) – (5.5) in the following chapter, and Fig.5.3 gives an overview over the whole audio codec classification stage, to classify AAC, AMR WB, and AMR NB coded audio content:



raw audio … unknown coded audio .wav content (AAC / AMR WB / AMR NB)

mcp ..... mean centre phase

mpr ..... mean phase range

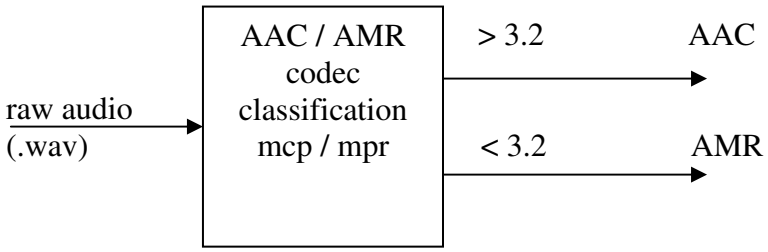
std (phase) ... standard deviation of the phase in rad

Figure 5.3: Audio codec classification stage of the reference free audio quality estimation system.

Equations for the mean centre phase, standard deviation of the phase, and mean phase range are given in the following chapter.

5.2.1 Reference free audio codec classification stage for AAC and AMR codecs

The reference free audio codec classification stage, the audio codec specific classifier, and the classification threshold value for classifying AAC and AMR codecs are shown in Fig.5.4:



raw audio .... unknown coded audio .wav content (AAC / AMR WB / AMR NB)

mcp / mpr ... mean centre phase to mean phase range ratio,

std (phase) ... standard deviation of the phase in rad.

Figure 5.4: Audio codec classification stage of the reference free audio quality estimation system.

In both audio codec classification stages, the unknown audio codec is classified by its codec depended characteristics. There can be several classifiers extracted from the frequency domain of an unknown coded audio content for audio codec detection, e.g., the cut-off frequency, the mean centre frequency or the mean bandwidth, reflecting the specific audio codec characteristics, depending on the individual performance audio codec settings by the user. A suitable audio codec classifier from the frequency domain for different coded audio content must classify the audio codec independently from the audio content, that means, it must classify all different kinds of coded audio content as AAC or AMR audio codec. Audio codec classifiers based on the cut-off and sampling frequency, mean centre frequency or mean bandwidth are not suitable for codec classification, while those parameter are depending on

the audio content and their values are very similar for AAC coded and AMR coded audio content. While different audio codecs use the same sampling frequencies and similar bitrates, audio codec classification based on such parameter will lead to no usable or significant classification results. Furthermore, some audio codecs support different bitrates at the same sampling frequency, so they cannot be classified by sampling frequency detection. Also, a bitrate detection for audio codec classification leads to no significant classification results, while, e.g., AAC works at 16kbit/s and AMR WB at 15.85kbit/s, both at the same sampling frequency of 16kHz. Another parameter that can be calculated from the mean centre frequency and the mean bandwidth of each frame is their ratio. Tests have shown, that this ratio mcf / mbw is usable for the classification of coded audio content, more exactly, to classify other music content from classic music content. Equation (5.1) represents the mean centre frequency and equation (5.3) gives an expression of the mean bandwidth:

$$\omega_c = \frac{\int_0^{\omega_0} \omega\,|F(\omega)|^2\,d\omega}{\int_0^{\omega_0} |F(\omega)|^2\,d\omega} \qquad (5.1)$$

where

ω_c … mean centre frequency mcf

ω … frequency

$F(\omega)$ … FFT coefficient

$$B^2 = \frac{\int_0^{\omega_0} (\omega - \omega_c)^2 \, |F(\omega)|^2 \, d\omega}{\int_0^{\omega_0} |F(\omega)|^2 \, d\omega} \qquad (5.2)$$

$$B = \sqrt{B} \qquad (5.3)$$

where

B … mean bandwidth mbw
ωc … mean centre frequency
ω … frequency
F(ω) … FFT coefficient

Further investigations have shown, that the mean ratio of the mean value and the mean range of the phase of each audio signal frame, similar calculated as the mean centre frequency and the mean bandwidth over all audio frames of an audio file, gives a significant classifier to separate AAC coded audio content from AMR coded audio content in a content independent way. The expressions for the mean centre phase and the mean phase range are given in expression 5.4 and 5.6:

$$mcp = \frac{\int_0^{\omega_0} \omega\,(phi(\omega))\,d\omega}{\int_0^{\omega_0} (phi(\omega))\,d\omega} \quad (5.4)$$

where

mcp … mean centre phase
ω … frequency
phi(ω) … phase

$$mpr^2 = \frac{\int_0^{\omega_0} (\omega - mcp)^2\,(phi(\omega))^2\,d\omega}{\int_0^{\omega_0} (phi(\omega))^2\,d\omega} \quad (5.5)$$

$$mpr = \sqrt{mpr} \quad (5.6)$$

where

mpr … mean phase range
ωc … mean centre frequency
ω … frequency
phi(ω) … phase at frequency ω

Fig.5.5 shows the classification results of this extracted classification parameter mcp / mpr for different AAC and AMR coded audio content:

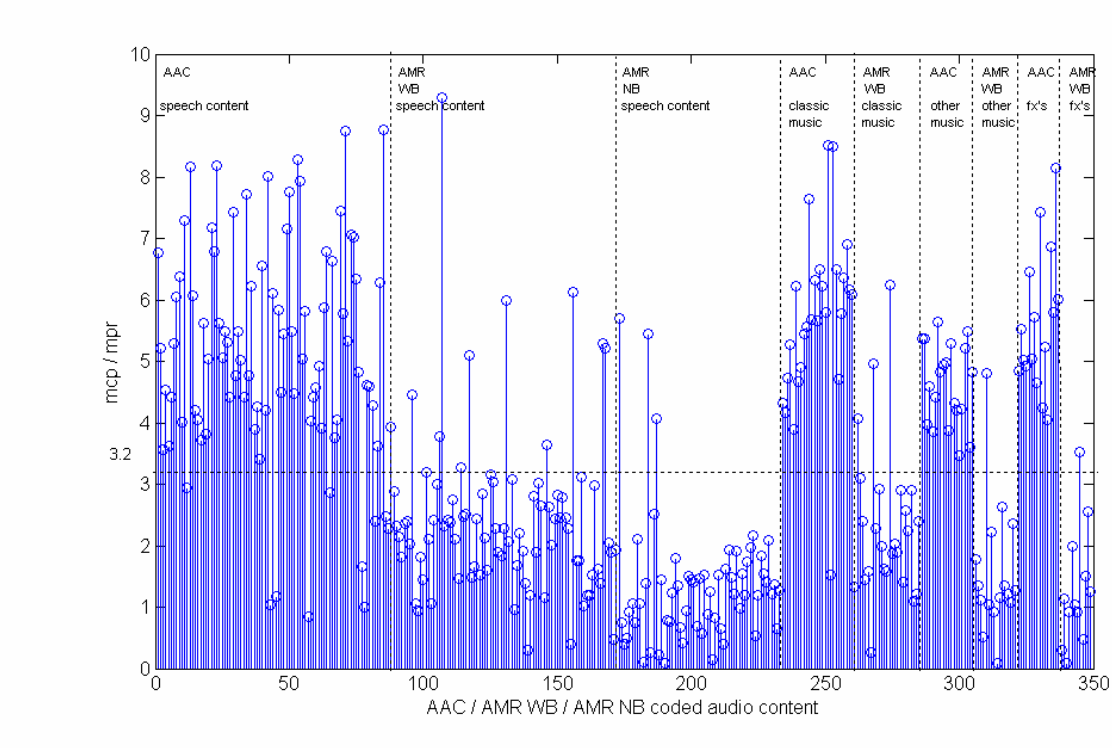


Figure 5.5: AAC / AMR audio codec classification results for different coded audio contents.

As described in chapter, the extracted parameter mean centre phase, called mcp, can also be used for AAC codec bitrate and sampling frequency classification and so, the mean centre phase mcp can be further seen as an optimal classification parameter in the whole audio codec characteristic classification system.

5.2.2 Reference free audio codec classification stage for AMR WB / AMR NB codecs

The optimal AAC and AMR classifier mcp / mpr cannot be used for AMR WB and AMR NB codec classification, as shown in Fig.5.5. One main difference between AMR WB and AMR

NB can be seen in the supported sample frequency or sample rate: AMR WB works for all bitrates at 16kHz and AMR NB at 8kHz. Once a parameter, reflecting the used sampling frequency, is found, it can be used as AMR WB and AMR NB codec classifier. While analyzing the phase of AMR WB and AMR NB coded audio signal, the standard deviation of the coded audio phase (std(phase)) in rad was found during tests as the most suitable parameter for classifying AMR WB coded audio content and AMR NB coded audio content. Equation (5.5) gives the expression of the standard deviation of the phase, while Fig.5.6 shows the AMR WB / AMR NB classification after the coded audio file was identified as AMR coded, while Fig.5.7 shows the AMR WB / AMR NB codec classification results for coded audio content speech, classic music, other music, and fx sounds:

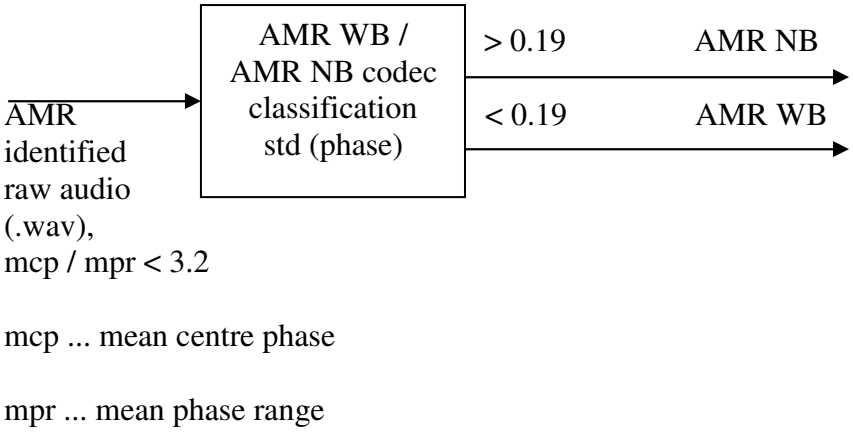


Figure 5.6: AMR WB / AMR NB codec classification stage.

Equations (5.7) and (5.8) give the expression of the standard deviation and mean value of the phase:

$$std(phi) = \left(1 / n \sum_{i=1}^{n} (phi_i - phi_m)^2 \right)^{0.5} \quad (5.7)$$

$$phi_m = 1 / n \sum_{i=1}^{n} (phi_i) \quad (5.8)$$

where

std (phi) … standard deviation of the phase in rad

phi_m … mean value of the phase

phi … phase

Once the coded audio content is classified as AMR, and the standard deviation of its phase is lower than 0.19, the audio codec is classified as AMR WB. Otherwise, it is classified as AMR NB. Furthermore, the standard deviation of the coded audio content phase in rad is also an optimal parameter: it can further be used to identify the sampling frequency of the unknown codec AMR WB or AMR NB. Classification results for AMR WB / AMR NB coded speech files, based on the standard deviation of the phase in rad, are shown in Fig.5.7:

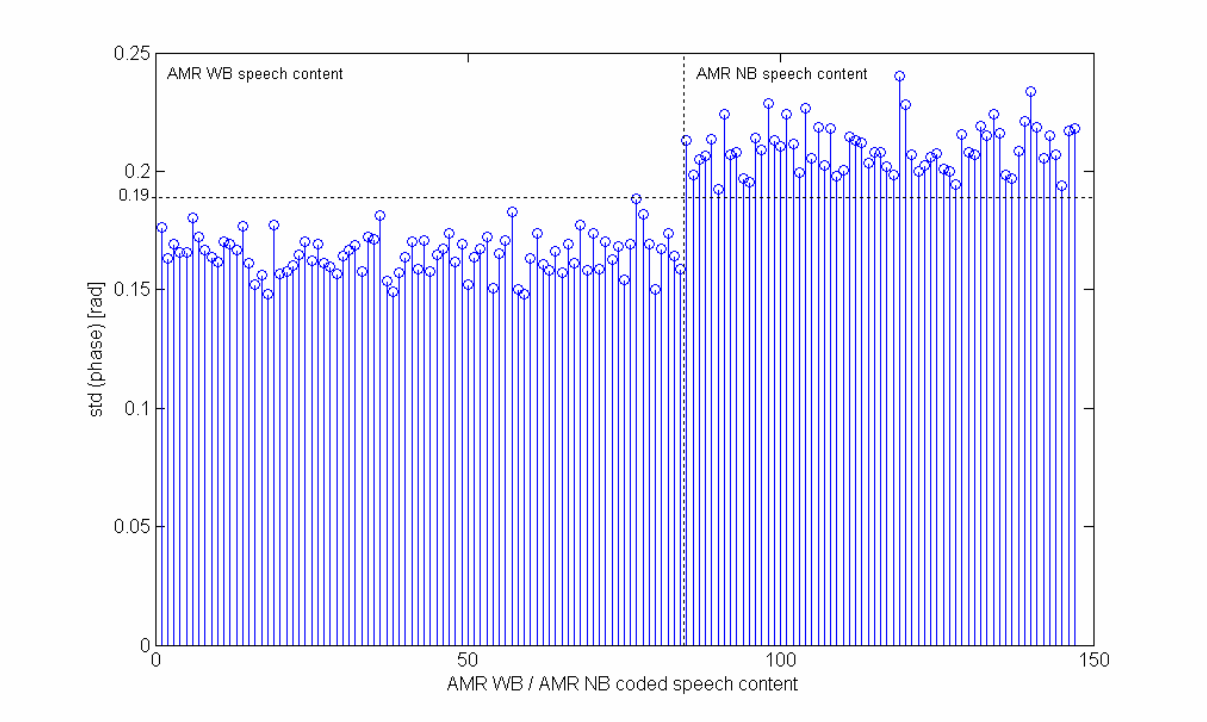


Figure 5.7: AMR WB / AMR NB codec classification results for different coded speech contents.

5.2.3 Reference free audio codec settings bitrate and sampling frequency classification stage

For the prediction of the bitrates and sampling frequencies, the available audio codec settings must be taken into account. For example, the sampling frequency of an AMR WB codec is 16kHz for every bitrate and 8kHz for every bitrate using AMR NB, while the sampling frequencies in case of AAC can be individually chosen for every bitrate. So, the audio codec classification of AMR WB or AMR NB determinates also the sampling frequencies, where in case of AAC, the specific chosen sampling frequency can be classified by the optimal parameter mean centre phase mcp, which is also used for the ratio of the mean centre phase and mean phase range for the classification of AAC and AMR.

5.2.3.1 Bitrate and sampling frequency classification for AAC coded audio content

The mean centre phase parameter mcp gives significant threshold values to classify the bitrate and sampling frequency of AAC coded audio content. Fig.5.8 shows this classifier for different audio content coded with AAC at different bitrates and sampling frequencies:

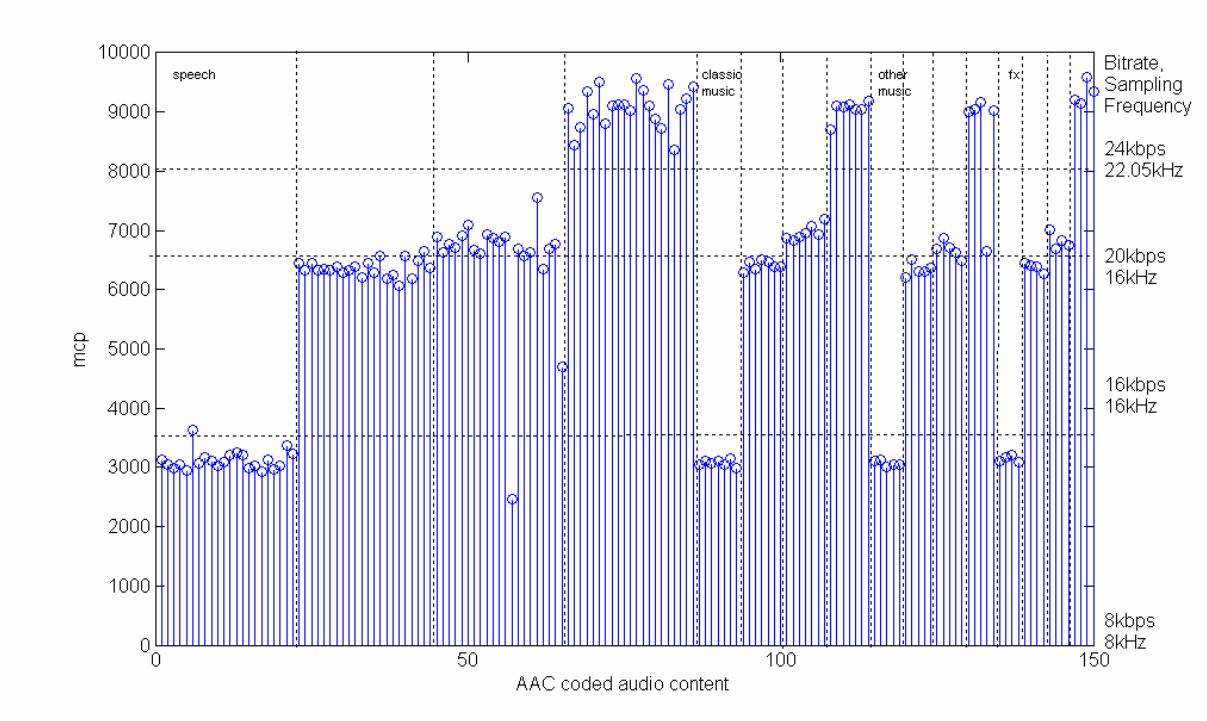


Figure 5.8: AAC bitrate and sampling frequency classification results for different coded audio contents.

5.2.3.2 Bitrate classification for AMR WB and AMR NB coded audio content

For AMR WB or AMR NB coded audio content, the mean centre phase mcp is not suitable for bitrate estimation. Typical audio codec bitrate characteristics were found in the combination of the standard deviation of the zero crossing rate and the mean value of the zero crossing rate, see equation (5.8) and equation (5.9). To use both parameter as AMR bitrate classifier, it was found, that the standard deviation and the mean value of the zero crossing rate of just the first 90 coded audio content samples were enough to classify the bitrates. Further, the threshold values for bitrate reconstruction or estimation are specific for each

audio content. Analyzing the standard deviation of the zero crossing rates, it was found, that for all audio codecs and audio contents there are only eight different values. Combined with other values of the mean of the zero crossing rate, estimation intervals for the original / recreated bitrate can be defined. Those eight different values of the standard deviation of the zero crossing rate are 0, 0.0047, 0.0094, 0.0141, 0.0236, 0.0189, 0.0283, 0.033 and were found empirically. For each value of the standard deviation of the zero crossing rate, the different mean values of the zero crossing rate can be divided into intervals, corresponding to equivalent bitrates. Therefore, it must be noticed, that one specific bitrate can correspond to more than one intervals and one specific value of a bitrate can be reached by several combinations of the standard and mean value of the zero crossing rate. For example, AMR WB 6.6kbps intervals for a standard deviation of the zero crossing rate equal 0.0189 are in the range of [0.14, 0.16] and [0.2, 0.29], interrupted by an AMR WB 15.85kbps interval, bounded at [0.17, 0.19]. A bitrate is also classified as AMR WB 6.6kbps, if the standard deviation of the zero crossing rate is 0.0094 and the mean value of the zero crossing rate is equal a value from the interval [0, 0.0067] or [0.199, 0.226]. Those bitrate classification combinations depend on the audio contents and Fig.5.8 gives two examples, how the bitrate for AMR WB coded speech can be reconstructed, if the standard deviation of the zero crossing rate of the modified audio content is equal 0.0283 or equal 0.0189:

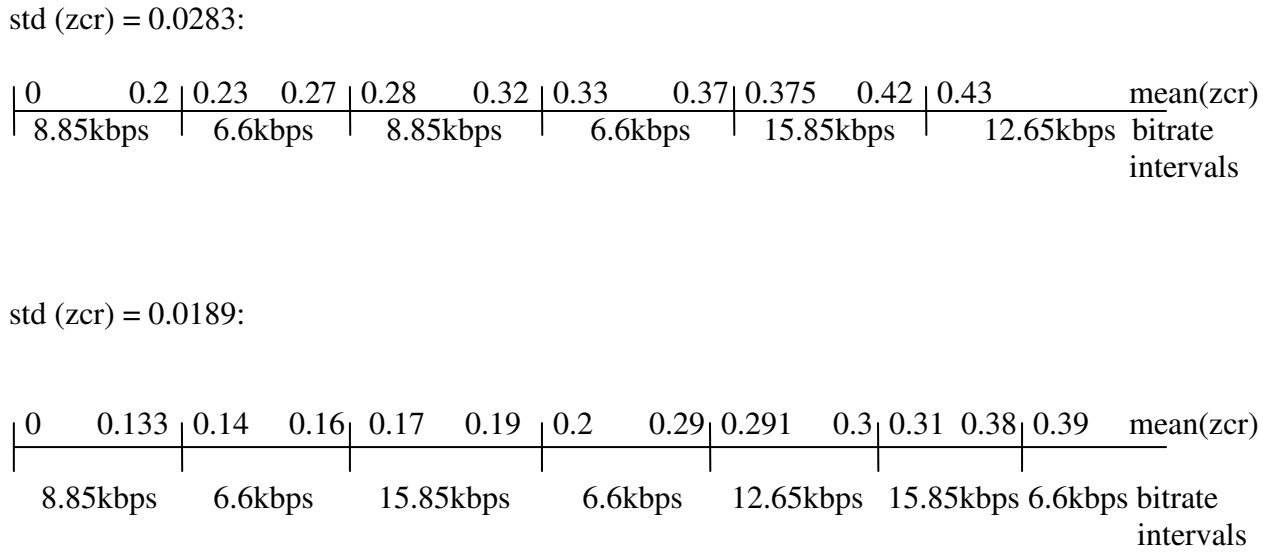


Figure 5.9: Examples for AMR WB bitrate estimation intervals.

Fig.5.9 shows, why the bitrate can only be classified by the combination of the standard deviation of the zero crossing rate and their mean value. A mean value of 0.34 can be caused by a bitrate of 6.6kbps or 15.85kbps. Together with the information about the standard deviation of the zero crossing rate, the bitrate can be correct classified as 6.6kbps or 15.85kbps. This example was chosen to show, that a mean value of the zero crossing rate within a specific interval corresponds to different bitrates, depending on the standard deviation of the zero crossing rate. Such bitrate classification intervals can be designed for each audio codec and audio content.

5.2.3.3 Sampling frequency classification for AMR WB and AMR NB coded audio content

One main difference between AAC coded audio content and AMR WB / AMR NB coded audio content is the constant codec specific sampling frequency of AMR WB and AMR NB. Once the audio codec is classified as AMR WB or AMR NB, the sampling frequencies for both codecs are given as 16kHz or 8kHz. Furthermore, the audio quality parameter coefficient c can also be used as an AMR WB and AMR NB sampling frequency classifier in combination with audio codec classification results. For example, if the audio quality parameter coefficient c of an unknown coded other music audio file is equal to 0.81 and the audio codec is classified as AMR WB, the only possible bitrate is 16kHz. A better AMR WB / AMR NB sampling frequency classifier is the standard deviation of the phase in rad, as mentioned in sub chapter 5.2.2. Again, the sampling frequency classification threshold to classify AMR WB and AMR NB is 0.19.

5.3 Reference free audio content classification

Audio content can be divided into two main content groups: speech content and non speech content. Further, non speech content can be divided into the sub categories music and fx sounds. Those subcategories can further be splitted into the different kinds of music or fx sound styles, as Fig.5.10 shows:

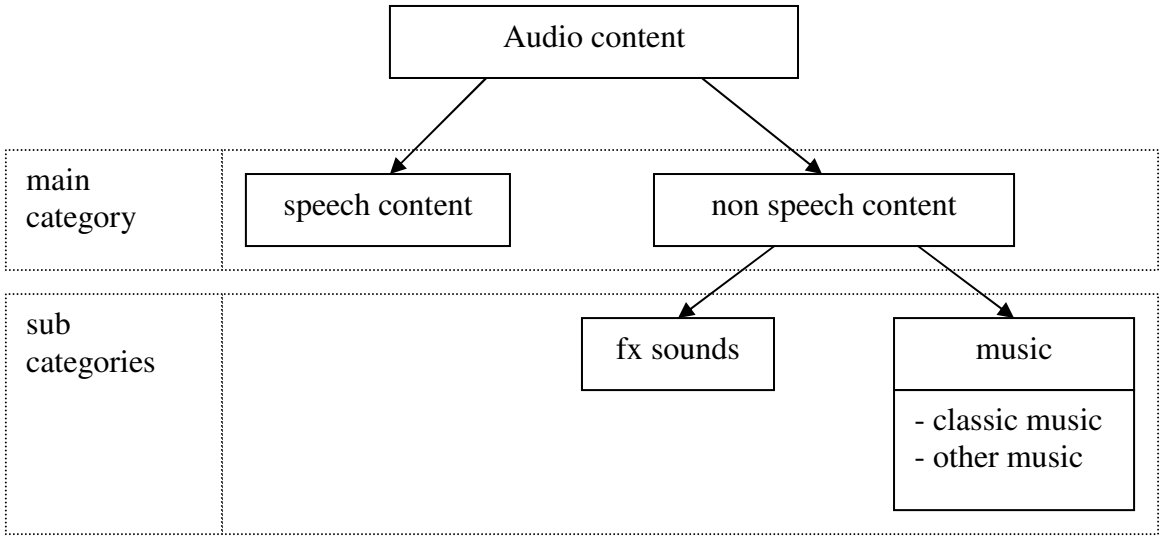
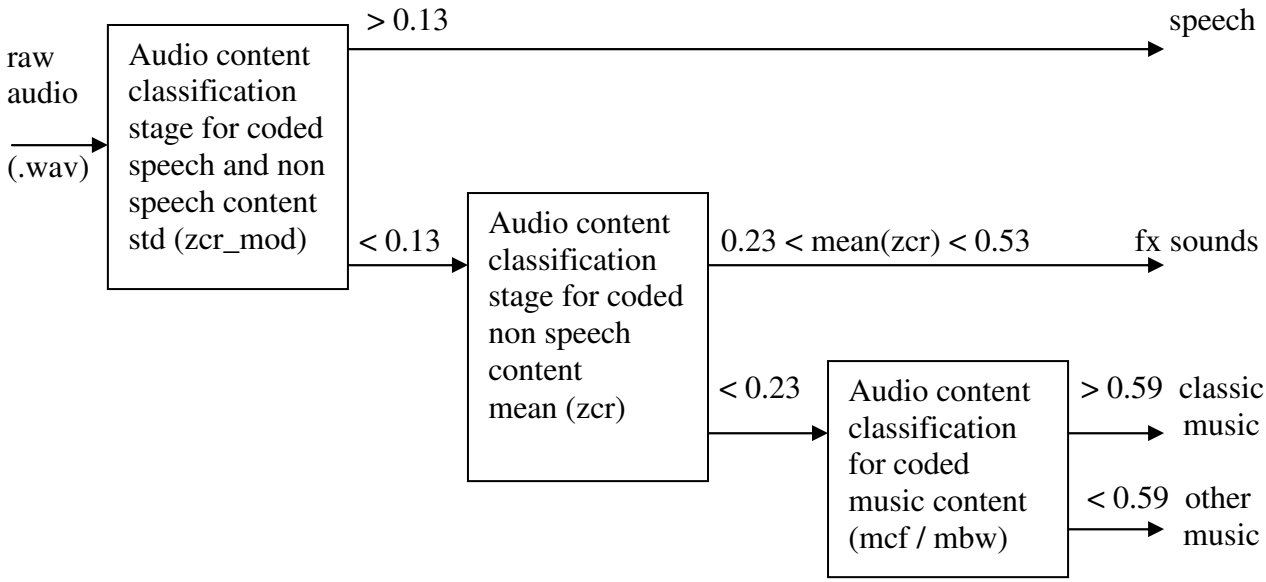


Figure 5.10: Audio content main- and sub categories.

The complexity of audio content classification systems grows with the number of different audio content sub categories. The following audio content classification system is developed for classifying the two main categories speech and non speech content in the first classification stage. Non speech content can be further divided into the sub categories classic music, other music and fx sounds. Fig.5.11 shows the whole audio content classification stage for coded audio content, while the expressions for music sub category classifier mcf / mbw are given in equation (5.1) and equation (5.3):



raw audio ..... unknown coded audio content (AAC / AMR WB /AMR NB)

std (zcr_mod) ..... standard deviation of the zero crossing rate of the modified audio file

mean (zcr_mod) ... mean value of the zero crossing rate of the modified audio file

mcf / mbw ..... mean centre frequency and mean bandwidth of the unmodified audio file

Figure 5.11: Audio content classification stage for coded audio content.

5.3.1 Reference free audio content classification for coded speech and non speech content

For developing an audio content classifier for coded audio content, based on time domain characteristics (zero crossing rate classifier), the audio codec specific characteristics in relation to their zero crossing points must be taken into account. To divide AAC from AMR coded audio content using zero crossing rate classification, a further audio signal forming process is necessary to make sure, that even AAC coded news speaker scenarios are classified as speech content. Audio content coded by AAC codec is characterized by extra samples introduced by this audio codec. Those samples influence the standard deviation of the zero crossing rate. So, an AAC coded speech file is characterized by a more periodic form of the zero crossing rate in a specific range and will be classified as non speech content. For a better

content classification of AAC coded speech files, the influence of the AAC encoder (extra samples) should be removed. Therefore, the locations of the extra samples within the audio file must be located for further signal processing operations, transforming those periodic patterns of the zero crossing points to non-periodic patterns. Fig.5.12 shows those extra samples of an AAC coded speech file (speech_stadt_aac_8.wav):

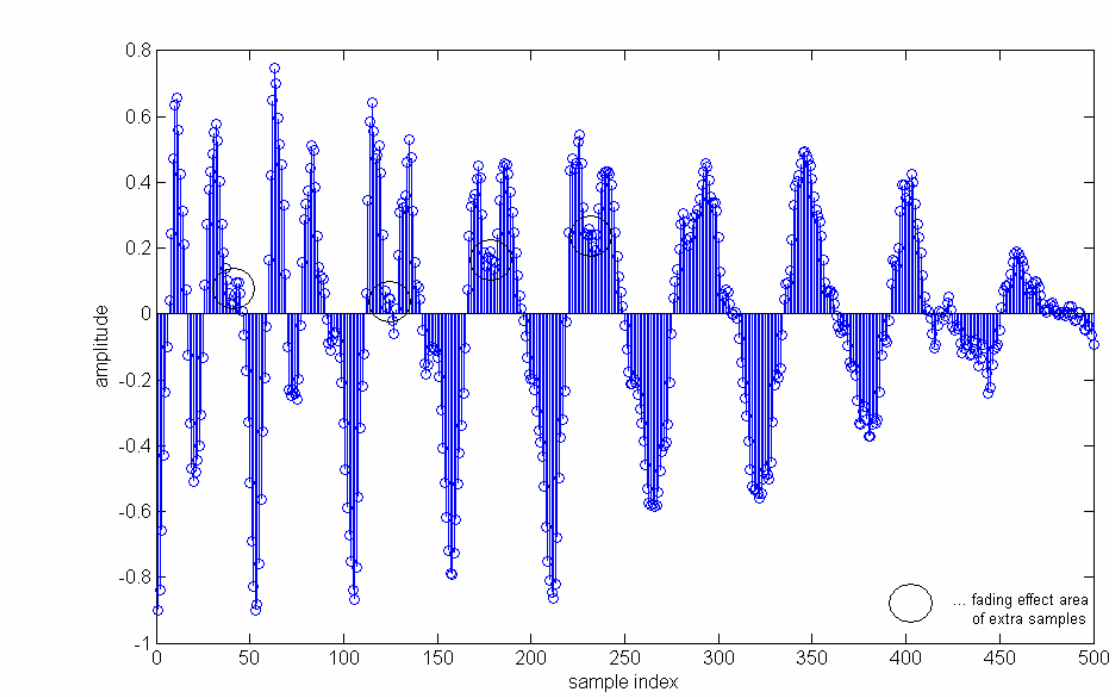


Figure 5.12: Influence of the AAC encoder in a speech file.

The amplitudes of those extra samples are very small, a fading effect over time can be noted and their zero crossing rate pattern seems to be periodic. To remove those AAC coding effects, all sample values lower than an empiric founded threshold are detected by an algorithm to set them to a specific value. Setting those samples just to zero has no significant influence on the final zero crossing rate and classification result. As mentioned above, a transforming of those periodic zero crossing patterns to non-periodic zero crossing patterns will satisfy the AAC coded speech content / non speech content classification. This pattern transformation is based on the following mechanism: the periodic structure of the zero crossing pattern (similar numbers of zero crossings in each frame over a specific range) is transformed randomly using the values of the neighbour samples of the extra samples, which are introduced by the AAC

encoder. While such time discrete signals also consists of negative sample values, a complex signal sample substitution at a specific time point solves the classification process of AAC coded speech best. The so modified AAC coded speech file, presented in the time domain, is given in Fig.5.13, while the transformation process is expressed in (5.10):

$$x(i) = \sqrt{(x(i-2))|} \qquad (5.10)$$
$$0 < x(i) < 0.1$$

where

x(i) … audio sample at time index i
x(i-2) … audio sample at time index i-2

Equation (5.10) shows, that the amplitude of the audio file at time point i is substituted by the value of the square root of the amplitude of the same audio file at time point i-2, if the amplitude at time point i is in the range [0, 0.1], which can also be seen in Fig. 5.13:

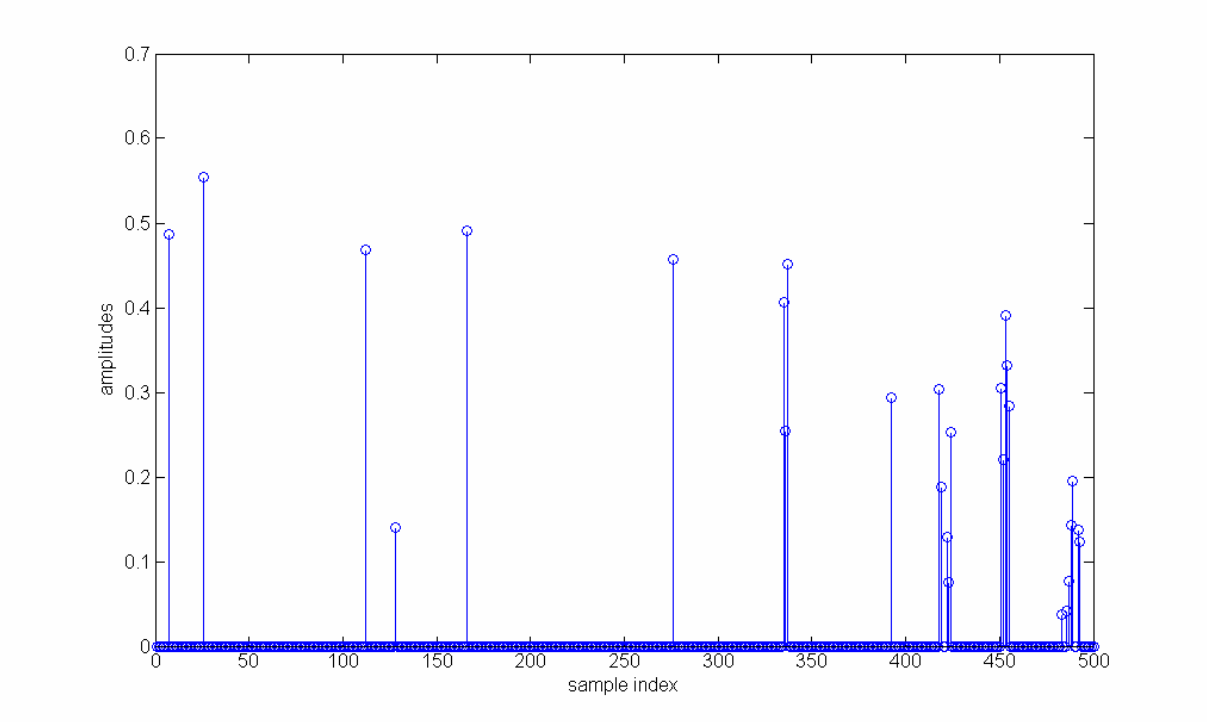


Figure 5.13: Modified speech signal in the time domain.

The number of zero crossing rates in each 150 sample frame of the original, unmodified AAC coded speech content file (speech_stadt.wav) with 16kbps, sampled at 16kHz, is shown in Fig. 5.14, and Fig.5.15 shows the number of zero crossing rates in each 150 sample frame of the modified AAC coded speech content file (speech_stadt.wav) with 16kbps, sampled at 16kHz:

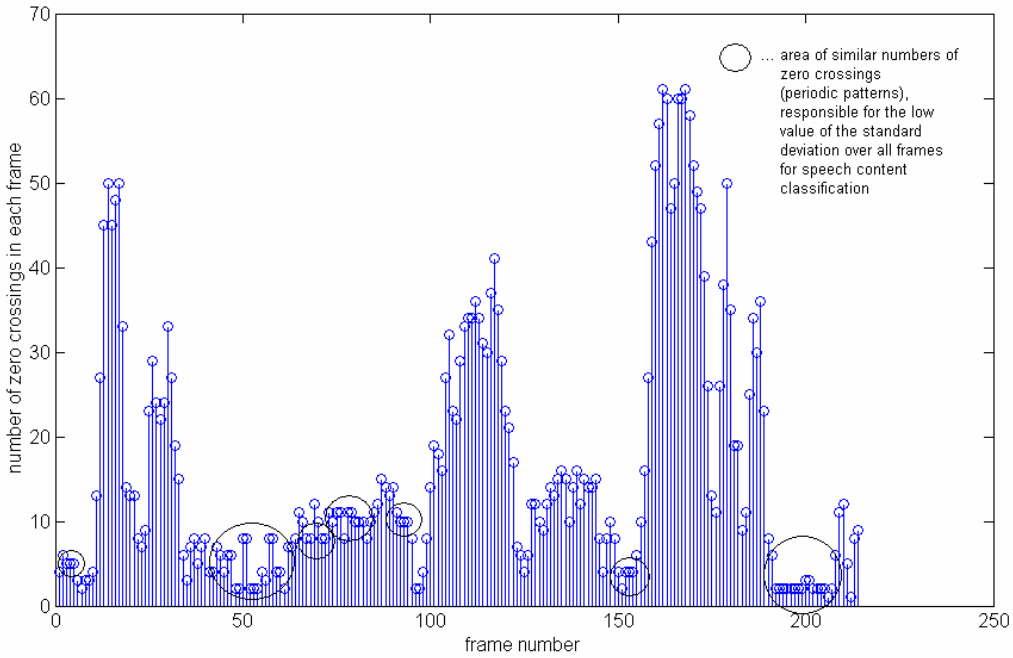


Figure 5.14: Zero crossings in each frame of unmodified, AAC coded speech file.

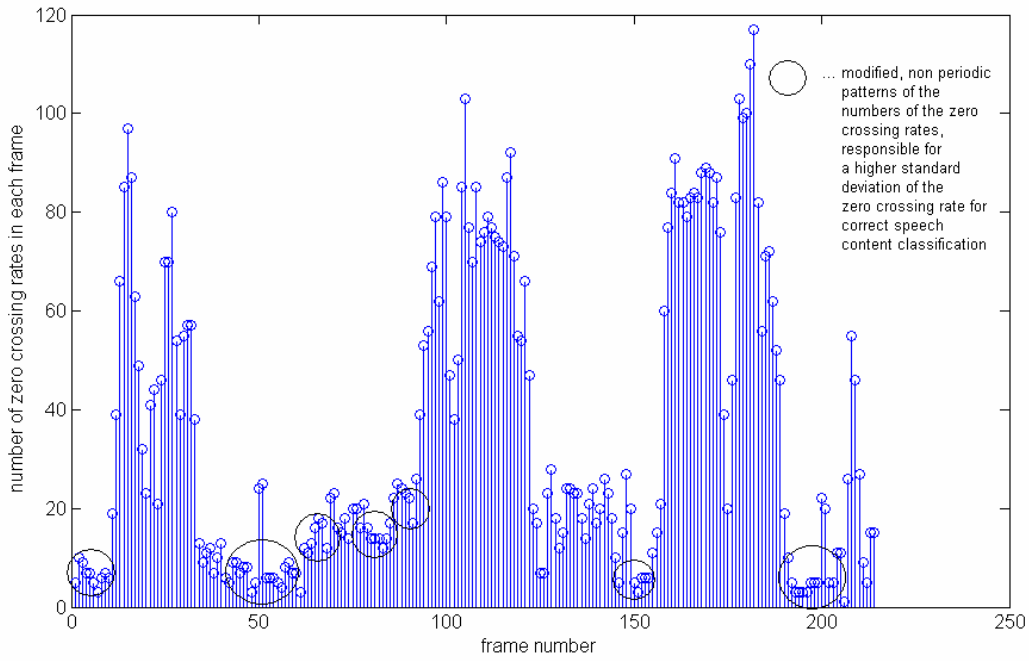


Figure 5.15: Zero crossings in each frame of modified, AAC coded speech file.

Comparing the results from Fig.5.14 with those from Fig.5.15, it can be seen, that the periodic pattern of the numbers of zero crossings for the case of the unmodified audio file are transformed to non periodic patterns in the case of the modified audio file, which leads to a higher standard deviation of the zero crossing rate resulting in a correct audio content classification for coded speech files. Further, comparing the results in Fig.5.14 and Fig.5.15, the numbers of the zero crossing rates in the marked areas of the modified audio file differs more from frame to frame, leading to a higher mean value of the standard deviation of the zero crossing rate over all frames than for the case of the original, unmodified audio file. Fig. 5.16 shows the number of zero crossings in each frame of a modified AMR WB coded speech file:

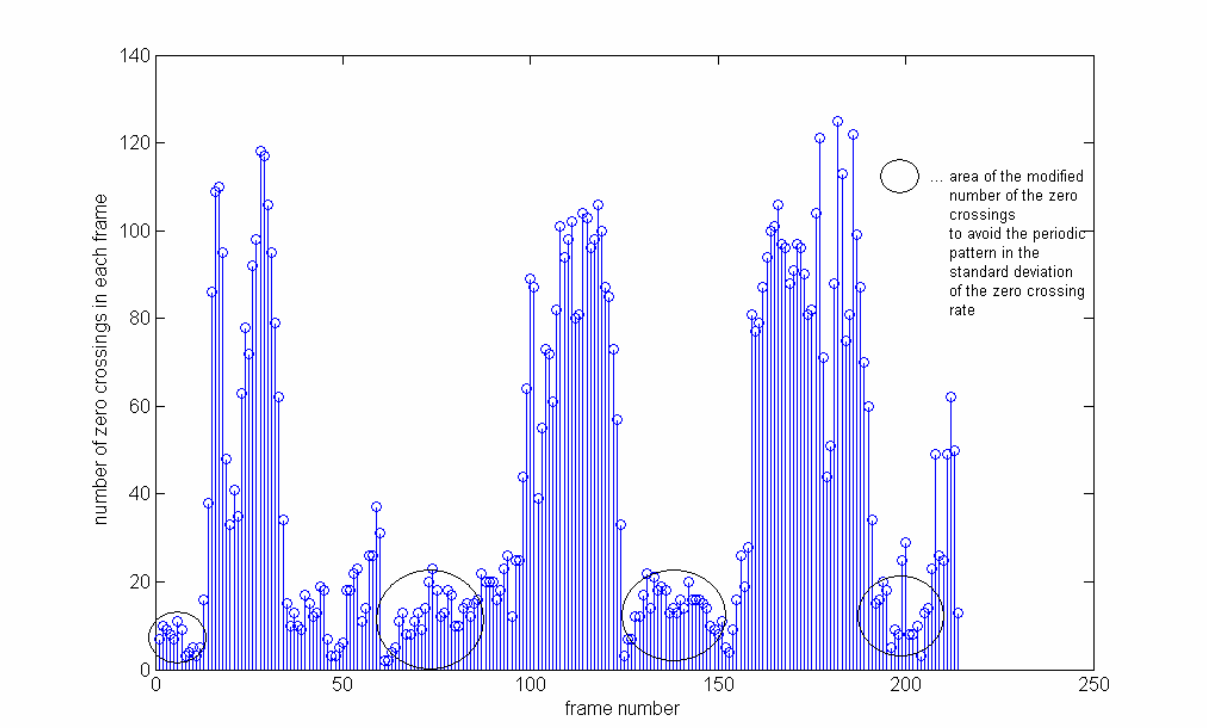


Figure 5.16: Number of zero crossings in each frame of a modified AMR WB coded speech file.

The following audio content classification process is done by this so transformed or modified audio signal, all other classification processes are based on the untransformed or unmodified audio signal. Using the mechanism from sub chapter 3.2 for uncoded audio content

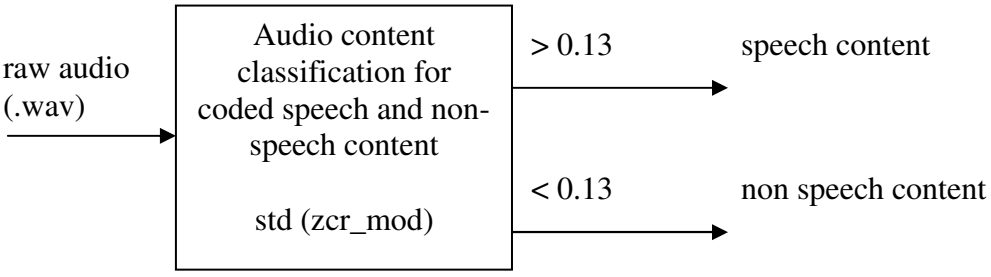
classification, the audio signal modification of the coded audio file and a threshold value variation of the standard deviation of the zero crossing rate to 0.13 enables the audio content classification of different coded audio content, codec independently.

An overview over the classification results of the unmodified and modified AAC coded speech file with codec settings 8kbps and 8kHz is given in Table 5.1:

AAC coded speech file	std (zcr)	classified as
Unmodified	0.0648	non speech content
Modified	0.1775	Speech content

Table 5.1: Audio content classification result for unmodified and modified AAC coded speech file

As Table 5.1 shows, the standard deviation of the unmodified AAC coded speech file with audio codec settings 16kbps and 16kHz is equal 0.0648 and would be classified as non speech content. The standard deviation of the modified AAC coded speech file with the same audio codec settings is equal 0.1775 and correctly classified as speech. This classification mechanism is shown in Fig.5.17, while Fig.5.18 shows the results of this classification for different coded speech and non speech content:



raw audio ... unknown coded audio content (AAC / AMR WB / AMR NB)
std (zcr_mod) ... standard deviation of the zero crossing rate of the modified audio file

Figure 5.17: Main audio content classification stage of the reference free audio quality estimation system for different coded audio content

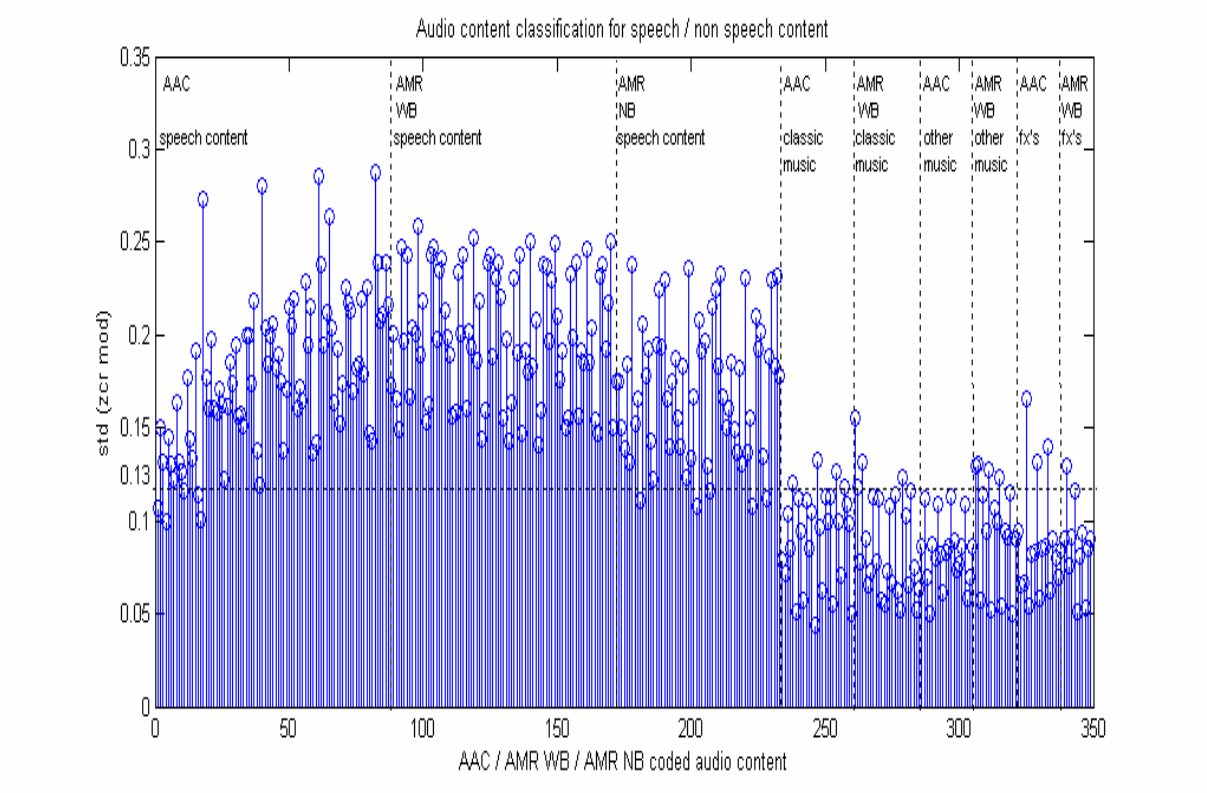


Figure 5.18: Main audio content classification results for different coded audio content.

Based on the detected audio codec information, the audio content classification consists of classification methods for each kind of audio content. While the standard deviation of the zero crossing rate of the transformed audio file classifies coded speech from coded non speech content, this parameter is not suitable for classifying different kinds of music content or fx sounds without further information. By combining this optimal parameter (standard deviation of the zero crossing rate) with the mean of the zero crossing rate mean (zcr_mod) and the mean centre frequency mean bandwidth ratio mcf / mbw of the unmodified audio file to a classification vector, it is possible to classify those three non speech sub categories. This extended classification mechanism for coded non speech content after the coded audio content was classified as non speech is shown in Fig.5.19:

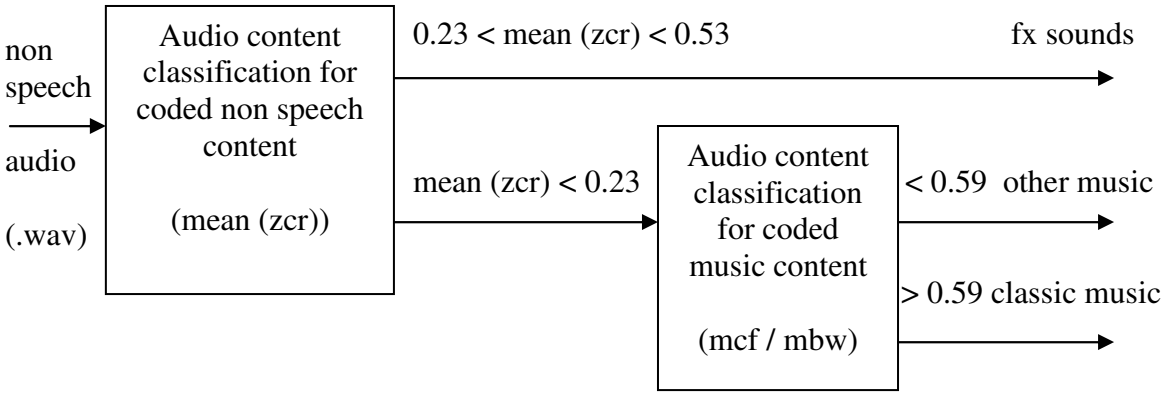


Figure 5.19: Sub category audio content classification stage for different coded audio content.

Results of coded fx sounds and music classification are shown in Fig.5.20, while Fig.5.21 illustrates the classification of other music and classic music:

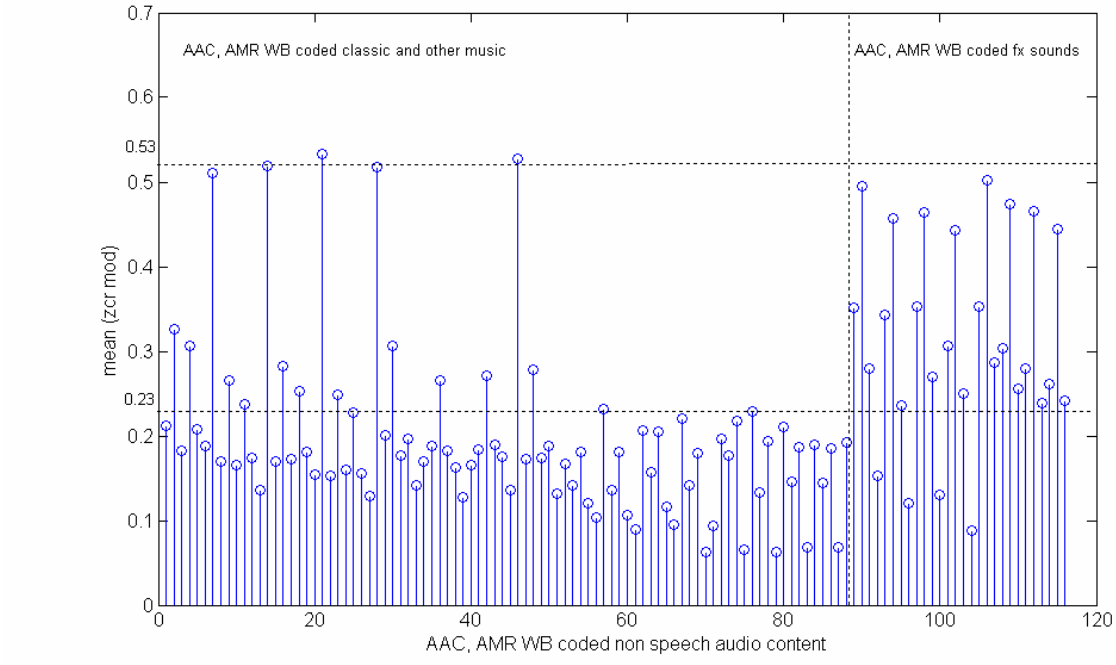


Figure 5.20: Results for subcategory music / fx sound content classification.

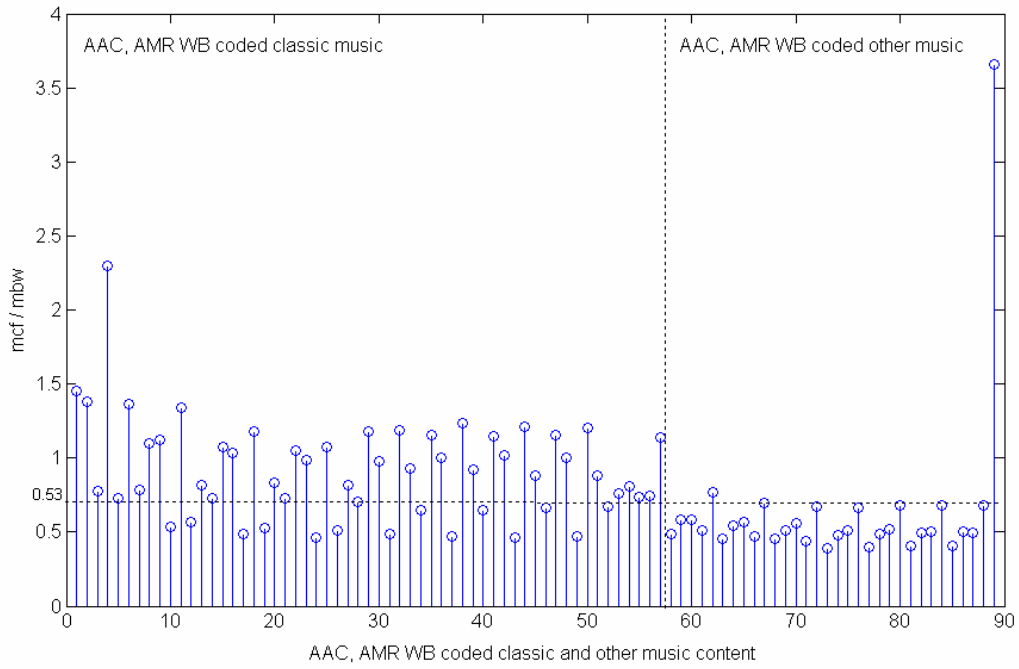


Figure 5.21: Results for subcategory classic music / other music content classification.

5.4 Reference free audio quality feature parameter extraction stage

For predicting the perceived audio quality of coded audio content based on automatically audio quality estimation metrics, a feature parameter unit extracts suitable audio quality feature parameter to design audio quality estimation metrics. The coefficients of the metrics can be further calculated by curve fitting, polynomial fitting algorithms or “linear in the parameter” regression. The complexity of those metrics depends on the used curve fitting algorithm, more exactly, on the order of the chosen approximation function and the number of different extracted feature parameter. Further, the different number of processing stages to find suitable feature parameters within the extraction unit, has a strong influence on the whole complexity of the system. So, lowest complexity of such a feature extraction unit in relation to computational power and calculation time is given, if a suitable metric parameter can be extracted by only one simple feature extraction stage without further signal processing units. As Fig.5.22 shows, the influence of the individual chosen audio codec performing settings bitrate and sample frequency is reflected by the codec specific deformations or attenuation of the magnitude and phase values of the coded audio file spectrum:

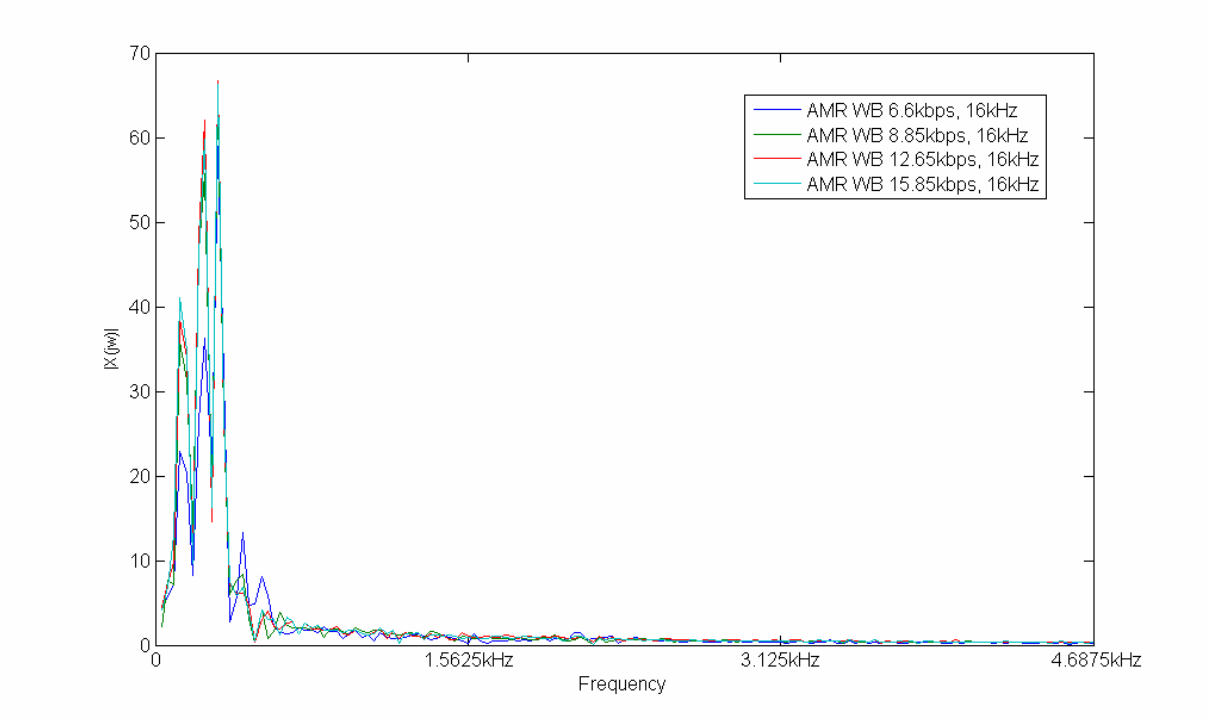


Figure 5.22: Frequency spectra of an AMR WB coded speech file for different bitrates.

From Fig.5.22, it is clear, that the deformation of the magnitudes in the frequency domain decreases with higher bitrates. So, higher bitrates can be associated with better audio quality. For example, the deformation or attenuation of AMR WB coded speech content at 6.6kbps is stronger than the deformation or attenuation of AMR WB coded speech content at 15.85kbps, both sampled at 16kHz. With the knowledge of how strong each audio codec deforms the spectrum of the coded audio file (mean magnitude over all audio frames) in relation to the results of the perceived audio quality listener tests (mean value of the subjective MOS scale values), it is possible to extract an audio quality feature parameter describing the influence of the audio codec settings on the perceived audio quality classification process of the test listeners. So, the reference information for predicting the audio quality of a coded audio file is given by the test results of the specific MOS test setup, reflecting directly the audio quality classification process of human beings, and must not be determed by the usage of other audio quality reference information as in case of feature extraction units based on perceptual model feature parameter extraction. In other words, the results of the subjective MOS listener tests are substituting the perceptual model unit in audio quality classification systems as a kind of universal reference source.

The influence of audio codec settings bitrate and sampling frequency (deformation, attenuation) on the audio quality of a coded audio file can also be shown in the time-frequency domain. Fig.5.23 gives an example of an original, uncoded classic music file, sampled at 44.1kHz and Fig.5.24 shows the same uncoded classic music file, sampled at 16kHz. While Fig.5.25 shows those influences on the audio quality for the same classic music file, coded with AMR WB 6.6kbps, 16kHz, and Fig.5.26 shows the time-frequency domain representation of the classic music file with codec settings 15.85kbps, 16kHz:

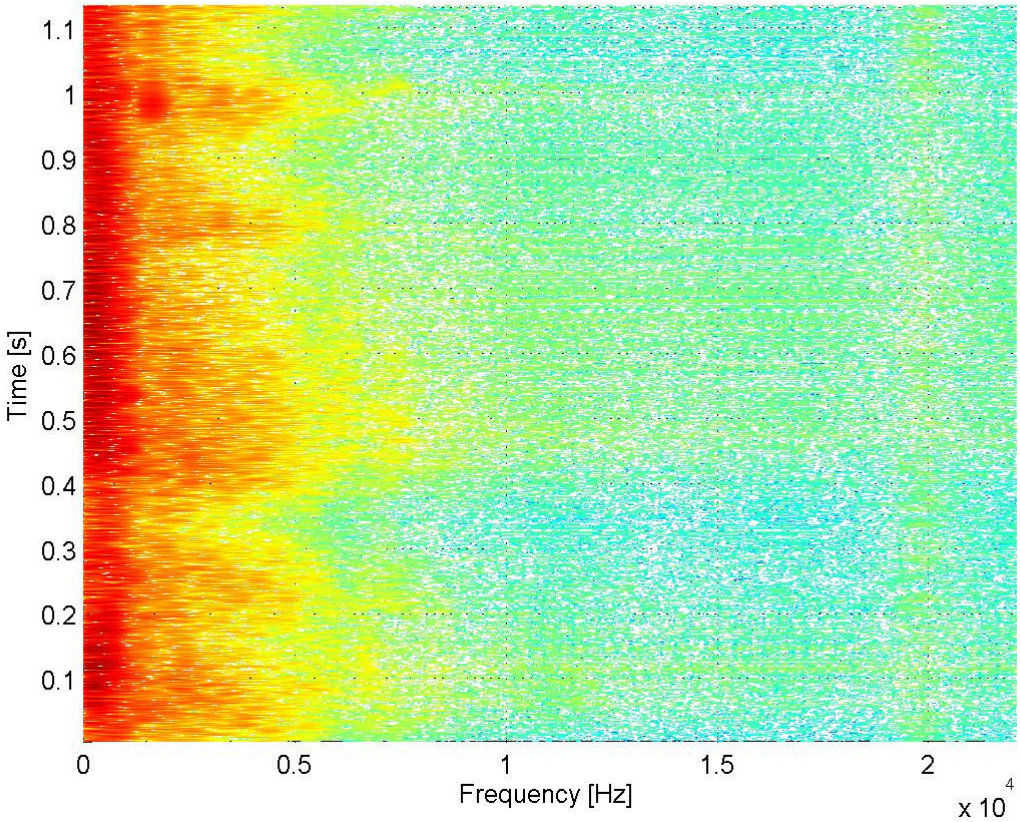


Figure 5.23: Time-Frequency domain of an uncoded classic music file, sampled at 44.1kHz.

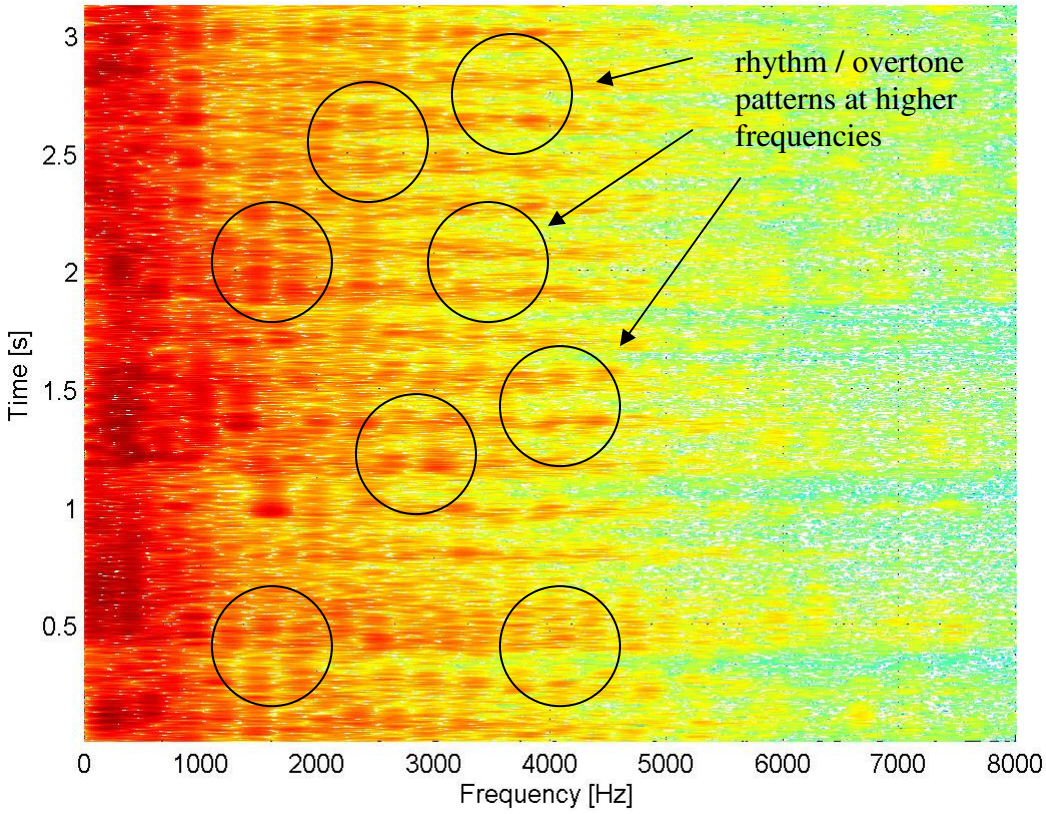


Figure 5.24: Time-Frequency domain of an uncoded classic music file, sampled at 16kHz.

Fig.5.23 shows typical periodic classic music rhythm / overtone patterns in the time-frequency domain of the original, uncoded audio file, sampled at 44.1kHz (high fidelity). Those rhythm / overtone patterns represent defined frequency groups over time intervals and those frequency groups can be interpreted as the basic tone and its harmonic overtones. In case of low audio quality, caused by audio codec settings bitrate equal 6.6kbit/s and sampling frequency 16kHz (Fig.5.25), the whole frequency groups (overtones) seem to be smeared in the frequency domain over a specific frequency range, caused by other frequencies appearing next to the harmonic overtones. Those smeared versions of the overtone patterns can be also seen as the spectrum deformation or magnitude attenuation as described for Fig.5.22, and further, that such audio codec settings do not preserve high frequencies, which are needed for high fidelity. Those extra frequencies are perceived as distortions and they influence the perceived audio quality. Further, those frequencies are caused by the audio codec settings bitrate and sampling frequencies, which are responsible for not preserving the high frequency areas, necessary for high fidelity, as shown in Fig.5.25 and Fig.5.26:

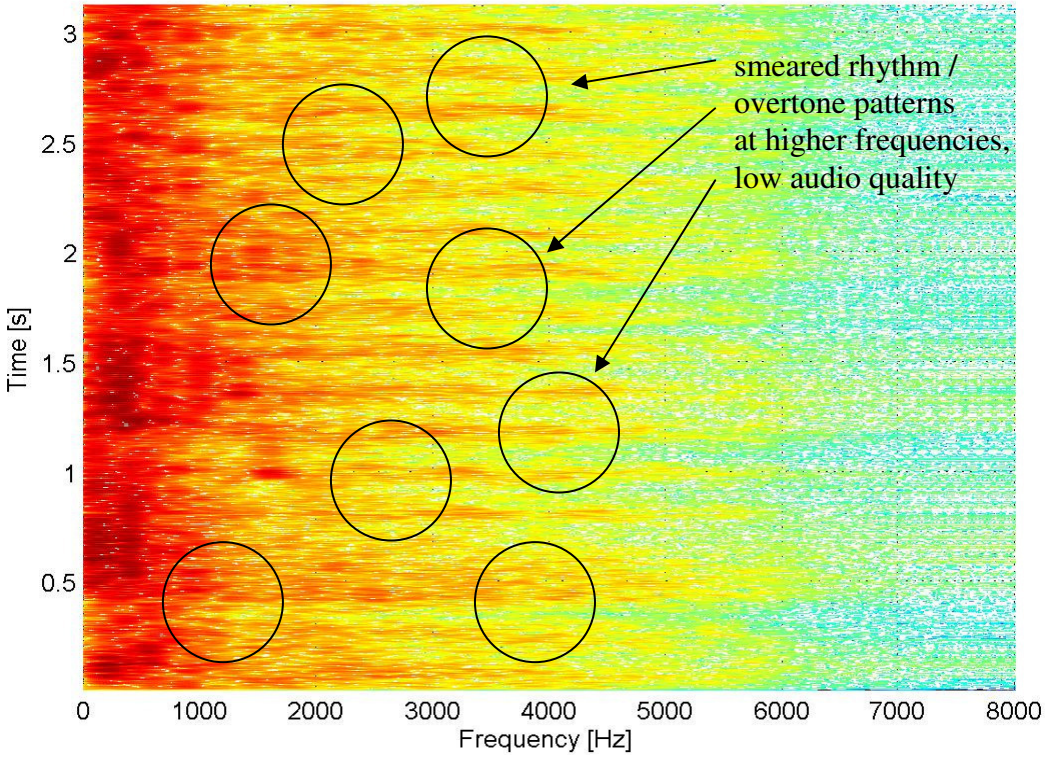


Figure 5.25: Time-Frequency domain of AMR WB coded classic music file, 6.6kbps, 16kHz.

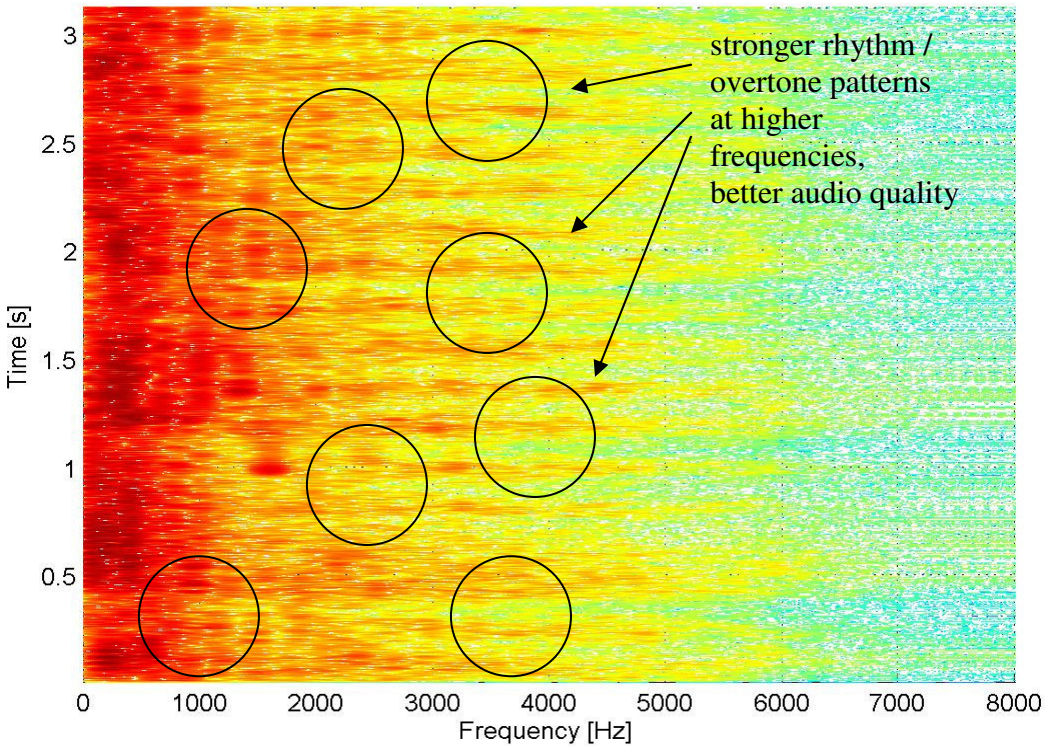


Figure 5.26: Time-Frequency domain of AMR WB coded classic music file, 15.85kbps, 16kHz.

The low audio quality of an AMR WB coded classic music with codec settings 6.6kbps, 16kHz can be seen in the smeared rhythm / overtone pattern in the time-frequency domain, while for AMR WB 15.85kbps, 16kHz, the ranges of those smeared rhythm patterns are smaller, leading to a higher audio quality. Finally, it can be said, that low audio quality corresponds with a strong smeared audio spectrum (strong deformation and attenuation), and higher audio quality with a lesser smeared audio spectrum (lower deformation and attenuation, higher frequencies are preserved).

For analyzing the frequency domain characteristic of time signals, first step in the signal processing algorithm is a Fast Fourier Transformation FFT, representing the time signal by its frequency domain characteristics magnitude and phase components. While all kind of audio signals are divided into frames of different length for audio signal processing and analyzing, mean values, like the mean magnitude, mean centre frequency or mean bandwidth over all frames of a coded audio file can be calculated and extracted as feature parameter, resulting in a single scalar value. The calculation complexity of suitable audio quality feature parameter grows then with the number of necessary further signal operation processes to extract them. For example, the feature extraction process of mean centre frequency and bandwidth is more complex than the feature process extracting the mean value of the magnitude of the spectrum of a coded audio file. This means, that the calculation of the mean centre frequency and mean bandwidth needs more calculation operations than the calculation of the mean magnitude. In that sense, the mean value of the magnitude over all frames of a coded audio file spectrum can be seen as an optimal audio quality feature parameter for audio quality metric design. Once such feature parameter, optimal or not, are found, they can be used as the parameter in audio quality estimation design. The simplest model for an audio quality estimation metric to predict the perceived audio quality of an unknown coded audio file without reference is a linear equation of the form

$$MOS_{Apred} = c \cdot p, \tag{5.1}$$

with

c ... audio quality parameter coefficient c
p ... audio quality feature parameter p

Furthermore, this model can be seen as a reduced audio quality estimation metric, consisting of only one audio feature parameter p and its audio quality parameter coefficient c.
This is possible, if an audio quality feature parameter can be found, that reflects the mean of the subjective MOS scale values in a linear way. Then, the audio quality metric can be reduced to an audio quality equation, consisting of only one optimal feature parameter of first order with just one parameter coefficient. While the left side of the equation MOS_{Apred} can be substituted by the mean value of the MOS scale value for each coded content type, given by the subjective MOS test results, one specific audio quality parameter coefficient for each kind of different coded audio content can be expressed by the following ratio:

$$c = \text{mean}\,(MOS_{subj}) / p \qquad (5.2)$$

5.4.1 Reference free audio quality parameter c to MOS scale value mapping unit

One specific audio quality parameter coefficient c can be seen as an element of an audio codec setting and audio content specific interval, where the interval bounds are given by the chosen audio codec settings bitrate and sampling frequency, depending on the audio content.
Those intervals are further used for mapping the audio quality parameter coefficient c to the rounded mean value of the corresponding subjective MOS. A test setup of different coded audio content, consisting of 349 different coded audio files, was used to find those significant interval bounds. Fig.5.27 gives an example for the mapping intervals of AAC coded speech content:

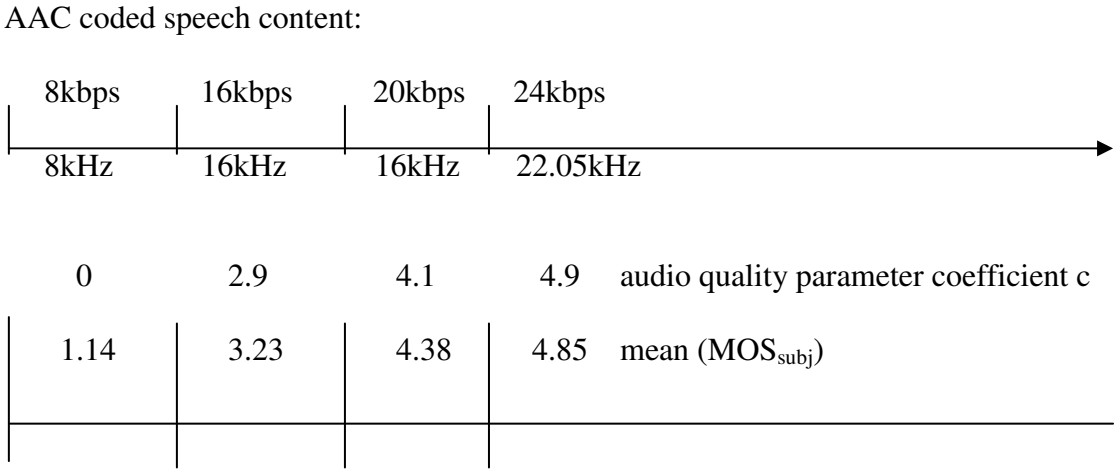


Figure 5.27: Example for an audio quality parameter coefficient c to mean (MOS_{subj}) mapping interval for AAC coded speech content.

For example, the audio quality parameter coefficients c values for AAC coded speech content with 16kbit/s, sampled at 16kHz, are distributed over an interval, bounded at [2.9, 4.1], and this interval is equal to the mean value of the subjective MOS (3.23), rounded to the next valid integer value 3. So, the audio quality parameter coefficient c is mapped to the integer value 3, representing the equal rounded, audio codec and audio content specific subjective MOS value. Table 5.2 shows the mapping interval bounds, the mean of the MOS value and its rounded version for each kind of audio codec, audio codec settings, and audio content:

audio codec	bitrate [kbit/s]	sampling frequency [kHz]	audio content	audio quality parameter coefficient c range	MOS_subj mean value	MOS scale value
AAC	8	8	speech	[0, 2.9]	1.14	1
AAC	16	16	speech	[2.9, 4.1]	3.23	3
AAC	20	16	speech	[4.1, 4.9]	4.38	4
AAC	24	22.05	speech	> 4.9	4.85	5
AMR WB	6.6	16	speech	[0, 1.9]	1.43	1
AMR WB	8.85	16	speech	[1.9, 3.9]	2.23	2

AMR WB	12.65	16	speech	[3.9, 4.5]	4.33	4
AMR WB	15.85	16	speech	> 4.9	4.81	5
AMR NB	4.75	16	speech	[0, 1.5]	1.14	1
AMR NB	7.95	8	speech	[1.5, 2.2]	1.91	2
AMR NB	12.2	8	speech	> 2.2	2.71	3
AAC	8	8	fx sounds	[0, 2.5]	1.47	1
AAC	16	16	fx sounds	[2.5, 3.1]	3.66	4
AAC	20	16	fx sounds	[3.1, 3.5]	4.43	4
AAC	24	22.05	fx sounds	> 3.5	4.66	5
AMR WB	6.6	16	fx sounds	[0, 0.9]	1.1	1
AMR WB	8.85	16	fx sounds	[0.9, 2]	1.476	2
AMR WB	12.65	16	fx sounds	[2, 2.52]	3.95	4
AMR WB	15.85	16	fx sounds	> 2.52	4.52	5
AAC	8	8	classic music	[0, 2]	1.66	2
AAC	16	16	classic music	[2, 2.8]	3.76	4
AAC	20	16	classic music	[2.8, 3.18]	4.66	5
AAC	24	22.05	classic music	> 3.18	4.95	5
AMR WB	6.6	16	classic music	[0, 0.81]	1	1
AMR WB	8.85	16	classic music	[0.81, 1.6]	1.1	1
AMR WB	12.65	16	classic music	[1.6, 2.36]	2.66	3
AMR WB	15.85	16	classic music	> 2.36	3.47	4
AAC	8	8	other music	[0, 1.5]	1.1	1
AAC	16	16	other music	[1.5, 2]	2.9	3
AAC	20	16	other music	[2, 3.5]	4.38	4
AAC	24	22.05	other music	> 3.5	4.85	5
AMR WB	6.6	16	other music	[0, 1]	1.095	1
AMR WB	8.85	16	other music	[1, 1.5]	1.38	1
AMR WB	12.65	16	other music	[1.5, 2.3]	2.85	3
AMR WB	15.85	16	other music	> 2.3	3.2	3

Table 5.2: Audio quality parameter coefficient c to MOS value mapping intervals for different coded audio content.

For the case of different fx sounds, a test setup number of different coded files lower than ten coded test files were enough to find significant classification bounds.

Examples of audio quality parameter coefficients c for different coded AAC speech files mapped to their corresponding MOS scale value are given in Table 5.3:

audio file name	audio codec	bitrate [kbit/s]	sampling frequency [kHz]	audio content	audio quality parameter coefficient c	mean value of MOS_{subj}	MOS scale value
speech_angel	AAC	8	8	speech	1.329	1.14	1
speech_cnn_1	AAC	8	8	speech	0.989	1.14	1
speech_eurosport_1	AAC	8	8	speech	1.584	1.14	1
speech_eurosport_2	AAC	8	8	speech	1.735	1.14	1
speech_matrix_1	AAC	8	8	speech	1.686	1.14	1
speech_angel	AAC	24	22.05	speech	6.087	4.85	5
speech_cnn_1	AAC	24	22.05	speech	4.677	4.85	5
speech_eurosport_1	AAC	24	22.05	speech	5.0144	4.85	5
speech_eurosport_2	AAC	24	22.05	speech	5.459	4.85	5
speech_matrix_1	AAC	24	22.05	speech	6.662	4.85	5

Table 5.3: Audio quality parameter coefficients c of different AAC coded speech files, mapped to their corresponding MOS values.

Table 5.3 shows, that the values of the audio quality parameter c and the mean value of the subjective MOS scale value can be mapped to the next nearest integer value by simply rounding functions. The results from the rounded audio quality parameter coefficient c and mean of the subjective MOS scale value differs only in one value, and so, the audio quality parameter coefficient c and its rounded version are suitable for predicting the perceived audio quality of different coded audio files of different audio content.

5.5 Reference free audio codec, audio codec settings, audio content, and audio quality estimation results

The whole audio quality estimation system was implemented in MATLAB (cf. appendix C). All MATLAB main programs and their syntax are given in Appendix C.3 - C.6 and the performance of the whole reference free audio quality estimation system, consisting of an audio codec classification stage, an audio codec setting estimation stage, an audio content classification stage, and an audio quality estimation stage was proofed by the audio test file setup given in Appendix C.2. All audio files (.wav) were chosen from audio codec setting specific folders with the following syntax:

audio_codec\audio_codec_bitrate\content_name.wav

For example, the syntax of an AMR WB coded speech file with the audio codec settings 6.6kbps and sampling frequency 16kHz is

amr_wb\6600\speech_stadt.wav

This means, that all AMR WB coded audio files with bitrate 6.6kbps are stored in a folder with path name amr_wb\6600\, all AMR WB coded audio files with bitrate 8.85kbps are stored in a folder with the path name amr_wb\8850\, and so on. Following this specification, it is possible to extract the whole information of audio codec, bitrate, and audio content from this text string. Also, parts of this information string can be extracted to compare the classification results with this kind of reference information about the test file. So, it is possible to give detail results of the whole audio quality estimation process, for example, the number of correct audio codec and audio bitrate classifications without information about the predicted audio content and audio quality.

5.5.1 Reference free audio codec, audio codec settings, audio content, and audio quality estimation results for unknown audio codec settings

Table 5.5 presents the precisions of each classification stage and Table 5.6 shows the results of the vector interpretation, both for 349 different coded audio files (cf. appendix C.2.1):

classification stage	number of correct classification	precision [%]
audio codec classification	326	93.41
audio codec bitrate classification	268	76.79
audio codec sampling frequency classification	347	99.43
audio content classification	288	82.52
audio quality estimation	246	70.49

Table 5.5: Precision of each classification stage for unknown audio codec settings.

vector representation	number of correct classification	precision [%]
Correct MOS scale value	246	70.49
+/- one MOS value precision	72	20.63
Correct MOS scale value and +/- one MOS value precision	318	91.12

Table 5.6: Correlation vector interpretation of the audio quality estimation results for 349 different coded audio files for unknown audio codec settings.

5.5.2 Detail results of audio codec, audio codec settings, and audio content classification

The following details of an unknown coded audio file can also be predicted by the audio quality estimation system:

- number of correct audio content and audio codec classification
- number of correct audio codec and audio codec bitrate classification
- number of correct audio content, audio codec, and audio codec bitrate classification
- number of correct audio content, audio codec, audio codec bitrate and sampling frequency classification

Table 5.7 gives an overview over the classification precision of those details:

classification of audio	number of correct classifications	precision [%]
Content and codec	265	75.93
codec and bitrate	268	76.79
Content, codec, and bitrate	221	63.32
Content, codec, bitrate, and sampling frequency	221	63.32

Table 5.7: Detail results of audio codec, audio codec settings, and audio content classification for unknown audio codec settings.

5.5.3 Reference free audio codec, audio content, and audio quality estimation results for known audio codec settings

The whole audio quality estimation system works also for known audio codec settings bitrate and sampling frequency. In that case, the information of the audio codec bitrates are extracted from the text string of the audio file. Table 5.8 presents the precisions of each classification stage for 349 different coded audio files and Table 5.9 shows the results of the vector interpretation:

classification stage	number of correct classification	precision [%]
audio codec classification	326	93.41
audio codec bitrate classification	349	100
audio codec sampling frequency classification	349	100
audio content classification	288	82.52
audio quality estimation	246	70.49

Table 5.8: Precision of each classification stage for known audio codec settings.

vector representation	number of correct lassification	precision [%]
Correct MOS scale value	246	70.49
+/- one MOS value precision	75	20.63
Correct MOS scale value and +/- one MOS value precision	321	91.12

Table 5.9: Correlation vector interpretation of the audio quality estimation results for 349 different coded audio files for known audio codec settings.

Comparing both audio quality estimation results, there are no significant differences in the estimation results, except in the results of the audio codec settings classification.

5.5.4 Detail results of audio codec, audio codec settings, audio content, and audio quality estimation

Table 5.10 gives an overview over the detail classification results:

classification of audio	number of correct classifications	precision [%]
content and codec	265	75.93
codec and bitrate	326	93.4
content, codec, and bitrate	265	75.93
content, codec, bitrate, and sampling frequency	264	75.64

Table 5.10: Detail results of audio codec, audio codec settings, and audio content classification for known audio codec settings.

5.6 Correlation between MOS_{Apred} and MOS from subjective listener tests

The correlation between the predicted MOS_{Apred} and the rounded mean value of the subjective MOS can be formulated by the Pearson linear correlation factor or by using a correlation vector. In case of the correlation vector interpretation or representation, the correlation between both MOS values can be described by the magnitude or phase of the correlation vector.

5.6.1 Correlation between MOS_{Apred} and MOS result from subjective listener tests, expressed by the Pearson linear correlation factor

The correlation between the mapped audio parameter coefficient c (predicted MOS_A) and the rounded version of the mean subjective MOS value can be expressed by the Pearson linear correlation factor:

$$r = \frac{x^T y}{\sqrt{(x^T x)(y^T y)}}$$

where, for the case of the audio test file setup described in appendix C.2,

x ... vector consisting of 349 predicted MOS_{Apred} values, specific for each codec and content

y ... vector consisting of 349 rounded mean values of MOS scale value from subjective listener tests, specific for each codec and content.

For the same test setup, consisting of 349 different coded audio files (cf. appendix C.2.1), the Pearson linear correlation factor is 0.867.

5.6.2 Correlation between MOS_{Apred} and MOS result from subjective listener tests, expressed by the components of a correlation vector (vector representation)

The rounded version of the mean subjective MOS value and the integer value of the predicted MOS, based on the mapped audio quality parameter coefficient c (MOS_{Apred}) can be represented on the axis of a coordinate system. Then, the vector represents a correct or wrong decision, by using the magnitude or phase of this correlation vector. In case of correct classification, the phase of the correlation vector to the correct classification line of 45° is zero. If the phase difference of the correlation vector to the correct classification line of 45° line is equal 15°, both MOS values differs in only one integer value. If the phase difference of the correlation vector to the correct classification line of 45° is equal 30°, both MOS values differs in two integer value. Fig.5.28 shows this correlation vector for a correct classification

and for the case of a predicted MOS_{Apred}, which differs in one MOS scale value in relation to a correct classification:

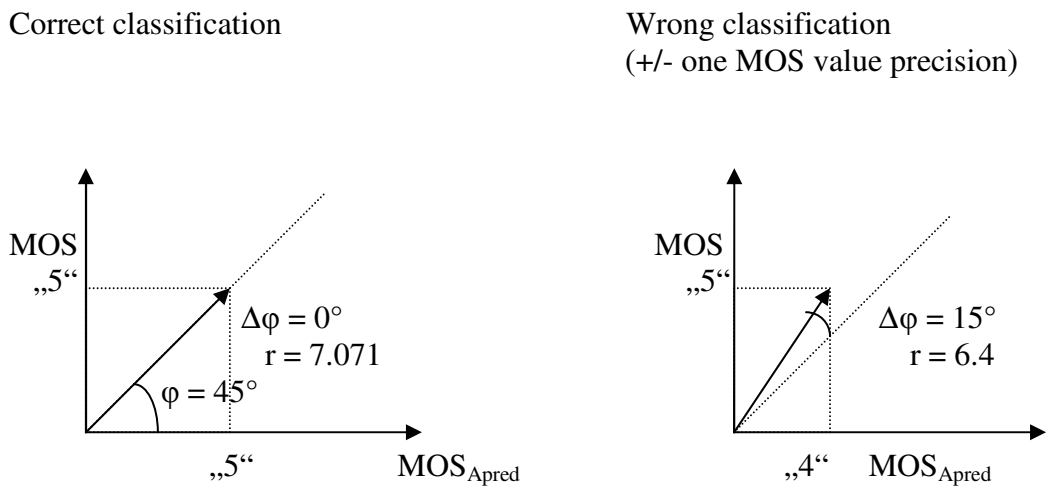


Figure 5.28: Vector representation for correct and wrong audio quality estimation using a correlation vector.

Identifying a correct classification with 100% (phase difference equal zero means no difference between both MOS values), then a phase difference of 15° is equal to a MOS difference to a correct classification of one integer value or MOS difference of 20%. 80% is identified with all correct predicted MOS_{Apred} values and those, which differs in only one integer value. By using the test setup of chapter, with this method of predicting the audio quality it is possible to reach 70.49% of 100% correct classification and 91.12% of 80%. Furthermore, this correlation vector representation is audio codec and audio content independent, and the values of the magnitude of the corresponding vector can be further used as an identifier for the correct MOS / MOS_{Apred} classification. Table 5.4 gives an overview over the magnitudes, the phase differences and the MOS values for the case of correct classification:

Magnitude (r)	Phase [°]	Phase difference to 45° line [°]	MOS_{Apred} = MOS	rated as
1.414	45	0	1	Bad
2.823	45	0	2	Poor
4.242	45	0	3	Fair
5.656	45	0	4	Good
7.071	45	0	5	Excellent

Table 5.4: Magnitude, phase, phase difference to 45° correct classification line, and MOS for correct audio quality estimations.

For example, if the magnitude (radius) is equal 7.071, the predicted MOS_{Apred} is equal the rounded version of the mean value of the subjective MOS, equal "5 ... excellent". If the magnitude (radius) is equal 1.42, the predicted MOS_A is equal the rounded version of the mean value of the subjective MOS, equal "1 ... bad". So, this identifier reflects both, the correctness of the classification and how a test listener would rate the audio quality of the file.

5.6 Classifier for reference free audio codec, audio codec settings, audio content, and audio quality estimation, conclusion

All of the classifier within the reference free audio quality estimation system and their threshold values are resumed in Table 5.5:

classification of	Classifier	threshold value
AAC / AMR	mcp / mpr	> 3.2 : AAC
AMR WB / AMR NB	std (phase)	> 0.19: AMR NB
AAC bitrate, sampling frequency	Mcp	see Fig. 5.8[1]
AMR bitrate, sampling freqency	mean(zcr), std(zcr)	combination of both[1]
speech / non speech	std (zcr_mod)	> 0.13: speech
fx sound / music	mean (zcr_mod)	0.23 < mean(zcr_mod) < 0.53
classic music / other music	mcf / mbw	> 0.59: classic music
MOS_{Apred}	round(mean(MOS_{subj})) mapped(mean(MOS_{subj})/p) mapped(c)	see Table 5.2[1]

Table 5.5: Classifiers and threshold values for reference free audio codec, audio codec settings, audio content, and audio quality estimation.

[1]) … classification based on more than one threshold values (intervals), of classifier combinations, or codec and content specific classification / mapping intervals.

Chapter 6

Reference free audio codec, audio content, and audio quality estimation for audio sequences

The reference free audio quality estimation system can be extended to predict the audio quality of each scene of a video sequence (video clip). Therefore, as described in chapter 3, the whole audio track is divided into single audio scenes, depending on the scene change cut time points or indexes, extracted by a scene detecting tool. This scene detecting tool is based on the scene change time points of the video sequence. So, the audio content, audio codec, the user specific audio codec settings bitrate and sampling frequency, and audio quality of each scene can be predicted. In a video streaming scenario, the whole audio track of a video clip is encoded by one specific audio codec with constant bitrate and sampling frequency. By using the first scene of the audio sequence, audio codec and bitrate of the whole sequence can be estimated.

- audio codec of the whole audio sequence
- audio codec bitrate of the whole audio sequence

Further, the following characteristics of each audio scene of the audio sequence are detectable:

- audio codec
- audio codec settings bitrate and sampling frequency
- audio content
- audio quality feature parameter p

- audio quality feature parameter coefficient c
- MOS_{Apred}
- Pearson correlation factor and correlation vector components
- processing time of each classification

The mean value of the audio content and MOS_{Apred} results of each scenes can be further used to predict the audio content and the overall MOS_{Apred} of the whole audio sequence.

6.1 Scene change detection tool for audio and video sequences

To estimate audio scene specific characteristics, such as audio codec, audio codec bitrates, sampling frequencies, and audio quality for an audio sequence extracted from a video sequence, it is necessary to detect the video scene change time points via a scene change detection tool to synchronize the audio scene changes. This video scene detection mechanism is described in the following section.

6.1.1 Video scene detection

Every video screen can be splitted into horizontal and vertical screen lines and every screen pixel contains information about its red, green, and blue intensity. A video scene change can be detected by a so called movement lattice, using the pixel colour information of specific chosen screen lines. By finding suitable relations between such extracted colour information from frame to frame, a scene change detection is possible. This video scene detection mechanism works as follows: the whole horizontal video screen size is reduced in that way, that only the colour information of the pixels of every tenth horizontal line is used for further processing. A value of ten was found empirically through several tests and the colour information for scene cut detection was chosen as red. The scene detection is based on the

following mechanism: first, a correlation coefficient vector is created, containing all correlation coefficients between the red colour intensity value of all successive frames. The correlation coefficients were chosen, while they give better information about scene changings in comparison to mean values or standard deviations of the red colour intensities, changing from frame to frame. Then, the difference between all those red colour intensity correlation coefficients are calculated. All red colour intensity correlation coefficient differences greater than an empirically founded threshold of 1.08 were combined to another vector. From that vector, again, the differences between two successive vector elements are created, and scene detection cuts were found on those locations, where those differences are greater than 2. Once those video scene change time points or indexes are detected, they can be transformed into the audio domain by the following equation:

$$n_{asi} = \text{round}\,(n_{vsi} \cdot F_s / \text{fps}) / 2 \qquad (6.1)$$

where

n_{asi} … new audio scene index
n_{vsi} … new video scene index
F_s … sampling frequency
fps … frames per second

So, the audio sequence is synchronized to the video scene change indexes, given at the scene change detection indexes nasi (new audio scene index). After each audio scenes are detected, their codecs, bitrates, sampling frequencies, contents and qualities can be estimated. To avoid lead in and lead out effects for AAC coded audio sequences, the first and the last extracted audio scene are not used for analyzation and so, the second audio scene is used for the audio codec and audio codec setting estimations of the whole audio sequence.

6.2 Reference free audio codec, audio content, and audio quality estimation for audio sequences, unknown audio codec settings bitrate and sampling frequencies

The following figures (Fig.6.1 – Fig.6.8) shows the classification results and classifiers of audio codec, audio codec bitrate, sampling frequency, audio content, and audio quality of each audio scene, extracted of a video sequence, consisting of three speech scenes and seven non speech scenes (classic music), encoded with AMR WB, with bitrate 15.85kbit/s and 16kHz, for the case of unknown audio codec, audio codec settings, and audio content:

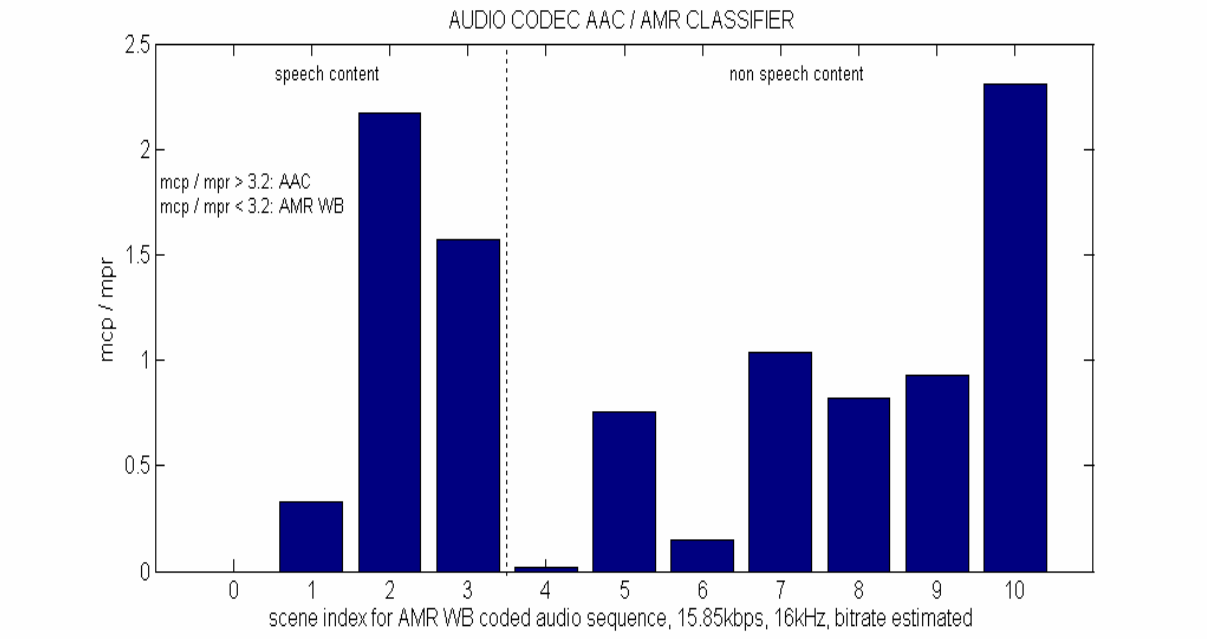


Figure 6.1: AAC / AMR audio codec classification results for each scene of an AMR WB coded audio sequence with different audio content.

As Fig.6.1 shows, all AAC / AMR classification ratios mcp / mpr of each scenes are lower than 3.2, and so, the audio codec of each scene is classified as AMR. The further classification of AMR WB and AMR NB and the bitrate classifiers are shown in Fig.6.2:

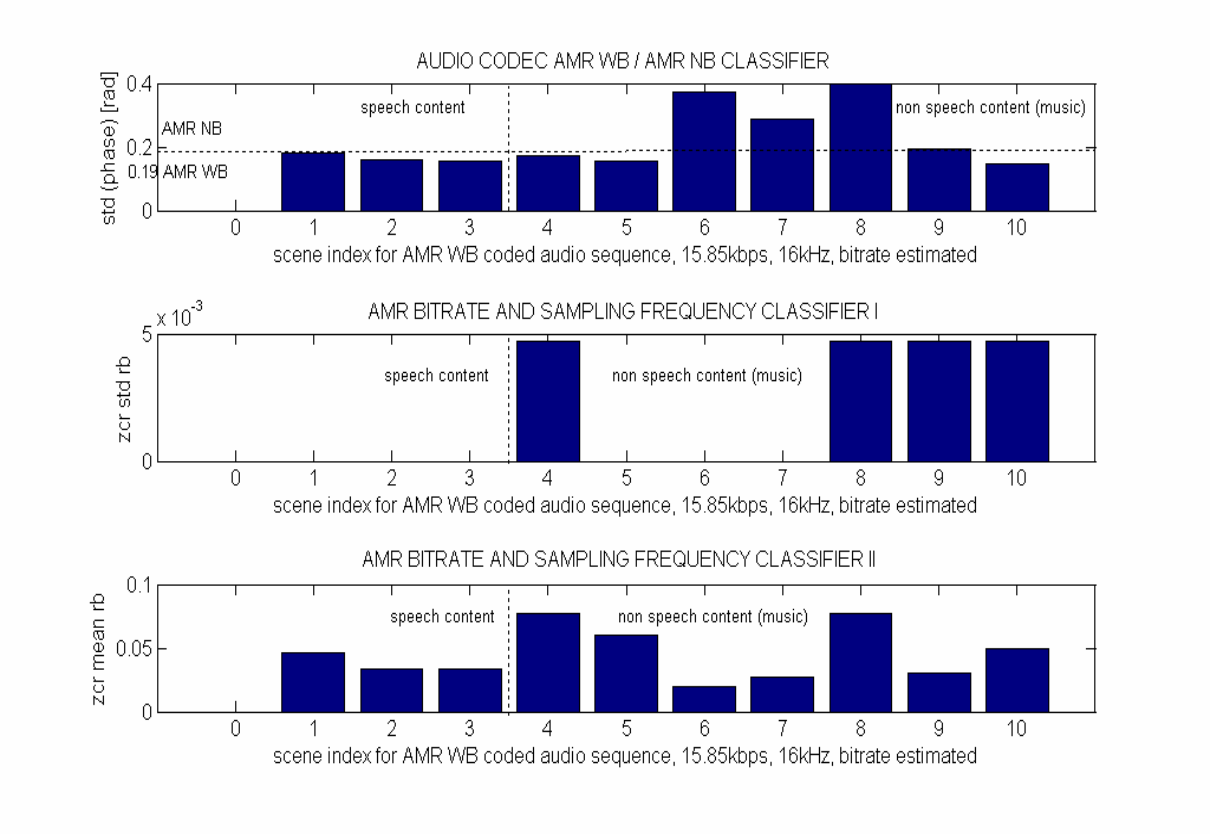


Figure 6.2: AMR WB / AMR NB codec classification, bitrate and sampling frequency classification results for each scene of an AMR WB coded audio sequence with different audio content.

The values of the standard deviations of the zero crossing rate for bitrate estimation of scene 1, 2, 3, 5, 6, 7 are not equal zero, they are only too low to be represented in the same diagram in relation to the standard deviations of scene 4, 8, 9, and 10.

Speech / non speech classification results of each scene, based on the modified audio file version, are shown in Fig. 6.3:

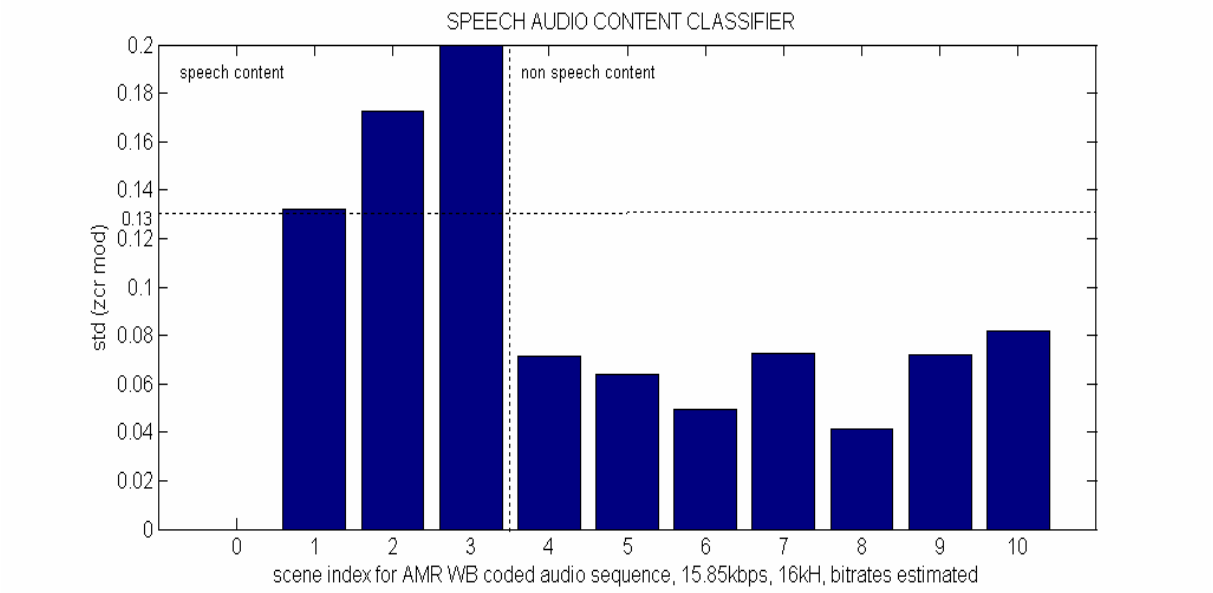


Figure 6.3: Main audio content classification results for each scene of the AMR WB coded audio sequence with different audio content.

As Fig.6.3 shows, the values of the standard deviation of the zero crossing rate of each modified audio file are upper than 0.13 for scene 1, scene 2, and scene 3, and so, they are classified correctly as speech content, while the values of the standard deviation of the zero crossing rate for the scenes 4 -10 are lower than 0.13, and so, they are correctly classified as non speech content. Once, an audio sequence is classified as speech, the mean value of the zero crossing rate and the mean centre frequency and mean bandwidth ratio (mcf / mbw) are irrelevant for further sub audio content classification. Those further coded audio content classifications to classify the sub categories fx sounds and music, classic music and other music, are shown in Fig.6.4:

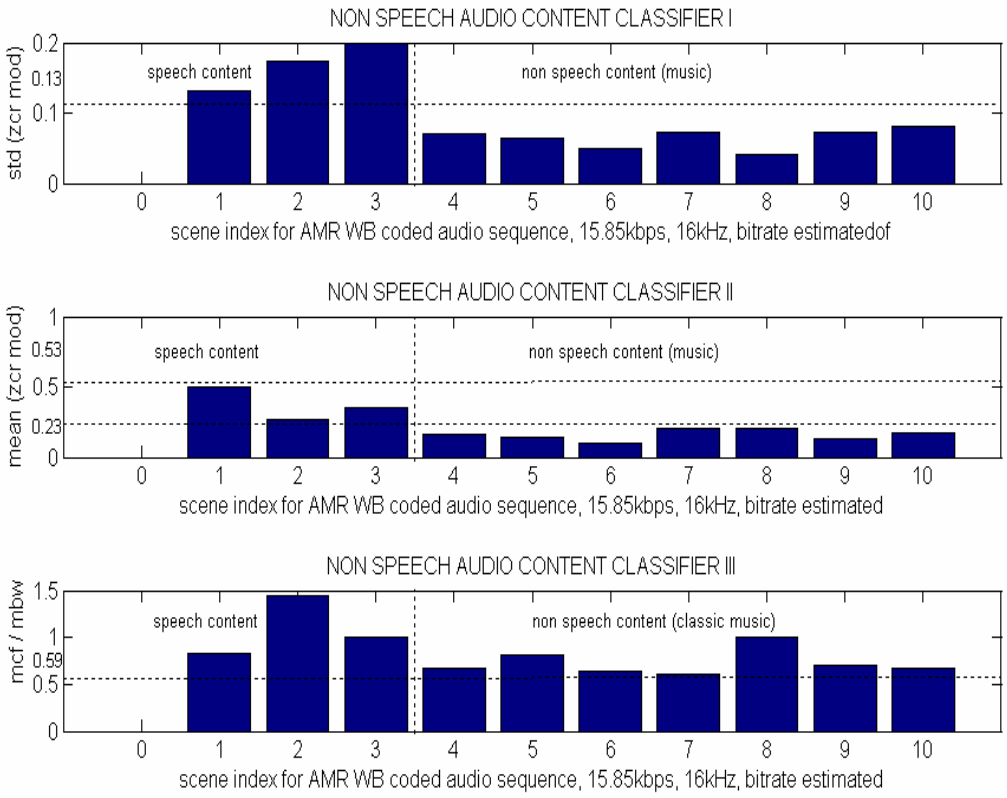


Figure 6.4: Main- and sub category audio content classification results for each scene of an AMR WB coded audio sequence with different audio content.

Non speech content is classified as fx sound, if the mean value of the zero crossing rate of the modified audio file is an element of the interval bounded at [0.23, 0.53]. If the values are lower than 0.23, the coded audio contents are classified as music. There is no need for further sub content classifications for scene 1, scene 2, and scene 3, while they are classified as speech, and the mean values of the zero crossing rates are not relevant. For example, scene 1 is not further classified as fx sound with a mean value of the zero crossing rate of 0.5.

The coded audio content classification triples for each scene are shown in Fig.6.5:

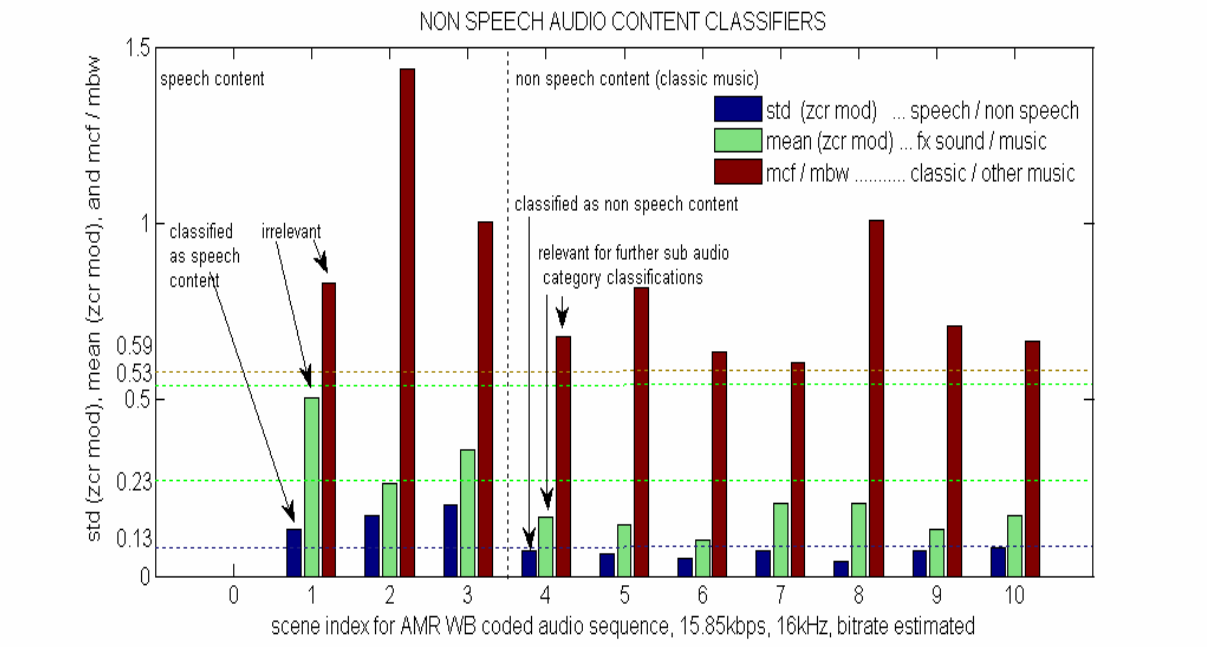


Figure 6.5: Audio content classification results for each scene of the AMR WB coded audio sequence with different audio content.

Once, a coded audio content is classified as speech, identified by a standard deviation value of the zero crossing rate upper than 0.13, no further sub audio content classification is necessary and the values of the two other classifiers (mean value of the zero crossing rate and mcf / mbw ratio) are irrelevant: this can be seen in the results of the coded audio content classification triple of scene 1, scene 2, scene 3. For all other scenes (scene 4 – scene 10), the standard deviation of the zero crossing rates are lower than 0.13, they are classified as non speech content, and so, the two other classifiers are relevant for further sub audio content classification. While none of the fx sound classifiers (mean value of the zero crossing rate) for scene 4 – scene 10 lies in the range of [0.23, 0.53], they are all classified as music. Further, while all their mcf / mbw ratios are upper than 0.59, all scenes (4 – 10) are correctly classified as classic music.

The relation between the audio quality parameter coefficient c, the estimated MOS_{Apred}, and the rounded mean value of the MOS value from subjective tests, audio codec settings and audio content specific, for each scene of the AMR WB coded audio sequence is shown in Fig.6.7:

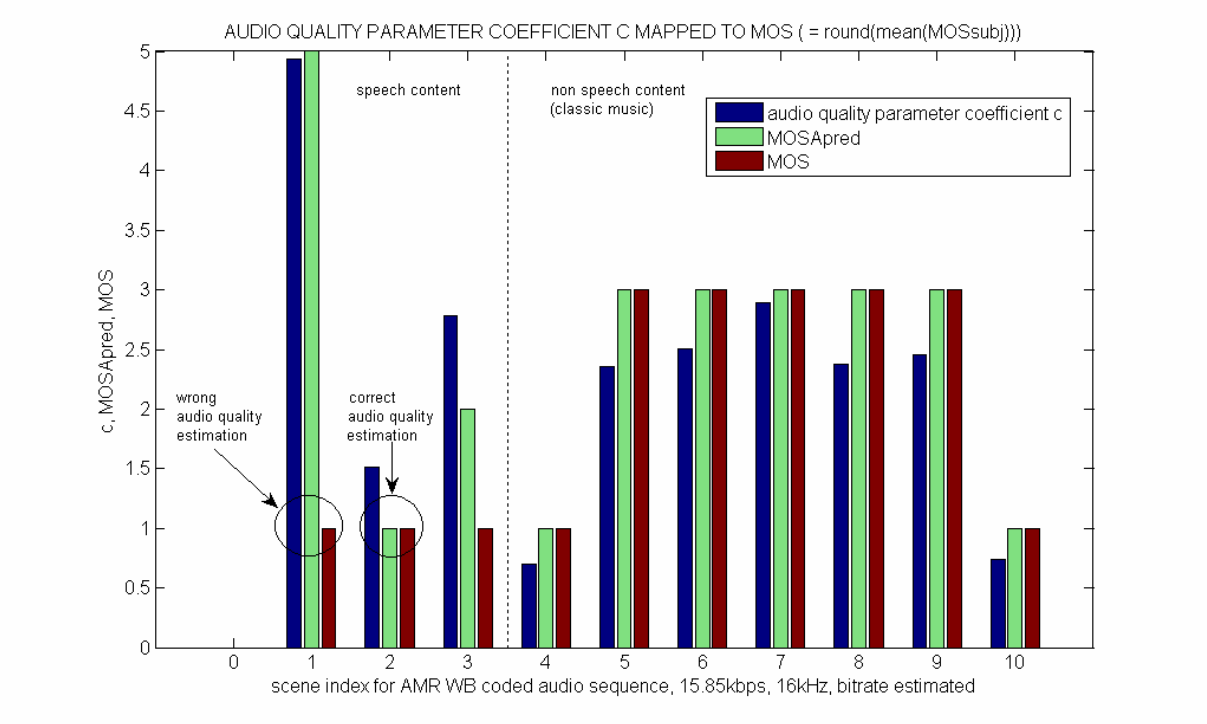


Figure 6.7: Audio quality estimation and mapping results for each scene of an AMR WB coded audio sequence with different audio content.

A wrong audio quality estimation result is shown in scene one, where the audio quality parameter coefficient c is mapped to a MOS_{Apred} value of 5, while the rounded mean value of the MOS scale value from subjective listener test is equal 1. Another wrong estimation result can be seen in the results for scene 3, while for all other scenes, the audio quality is estimated correctly (MOS_{Apred} equal MOS). The bitrates of scene 4 and scene 10 are not correctly estimated equal 15.85kbps, they are classified as 6.6kbps, but the predicted MOS_{Apred} is correctly estimated for this bitrate value, and so, it can be seen as a kind of correct estimation. The audio quality of the scenes 4 – scenes 9 are estimated correctly.

Audio quality estimation results for each scene, represented by their correlation vector components magnitude, phase, and phase difference to 45° right classification line are shown in Fig.6.8:

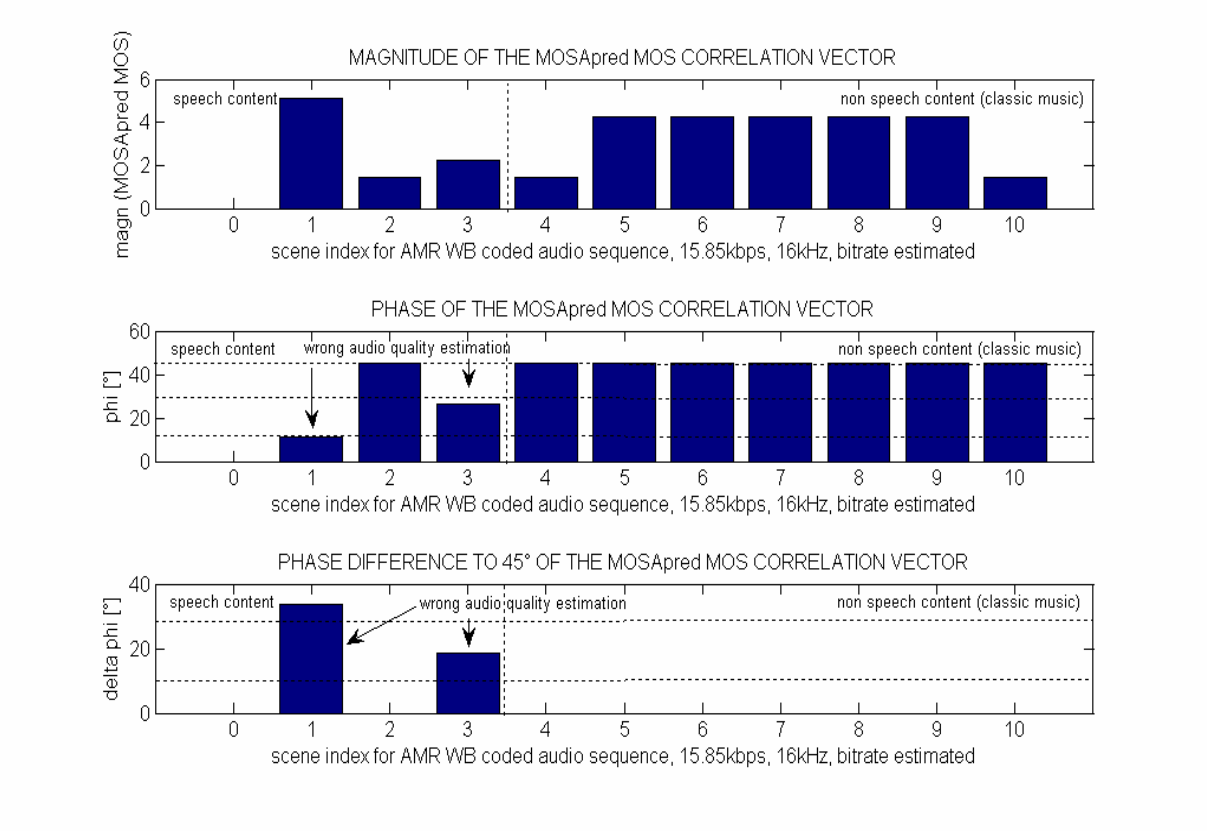


Figure 6.8: Audio quality estimation and mapping results for each scene of the audio sequence, correlation vector representation.

A wrong audio quality estimation result can be seen in the values of the correlation vector phase or in the correlation vector phase difference to 45° (scene 1, scene 3), whereas the audio quality of all other scenes are estimated correctly.

Program outputs and results for the audio quality estimation of the whole audio sequence and for one specific audio scene with unknown audio codec settings are given in appendix C.

6.3 Reference free audio codec, audio content and audio quality estimation for audio sequences, known audio codec settings

Classification results for coded audio content and audio quality estimation for each scene of the AMR WB coded audio sequence, coded with 15.85kbps, sampled at 16kHz, for known audio codec settings bitrate and sampling frequency are shown in Fig.6.9 - Fig.6.11:

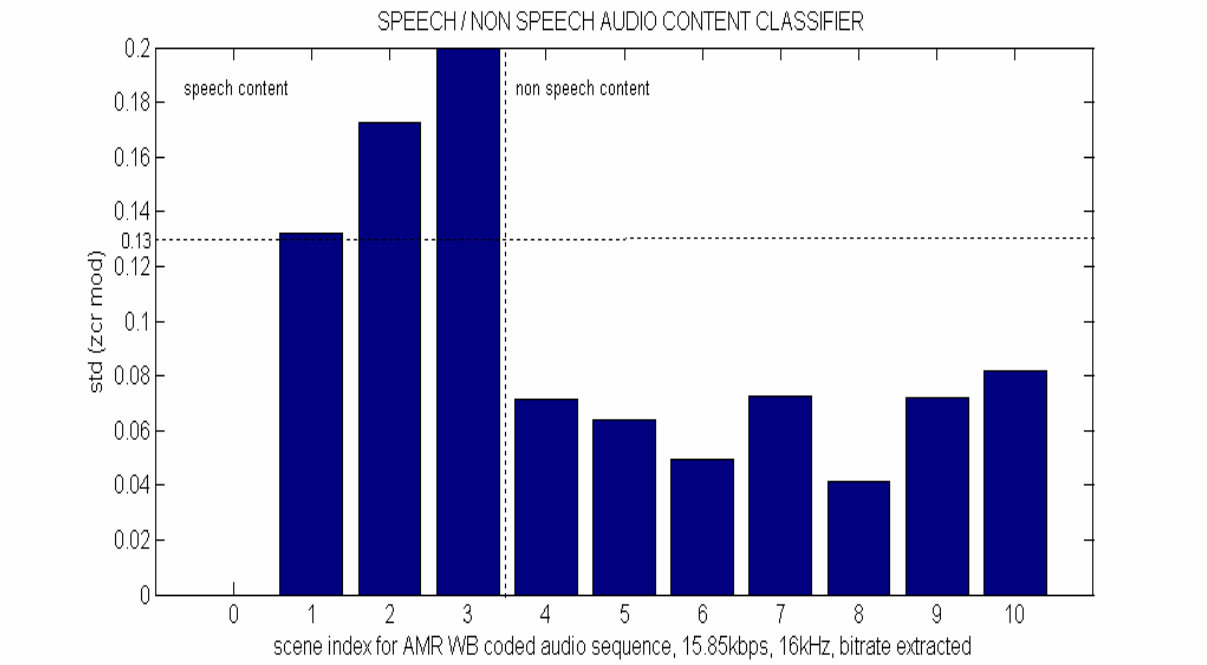


Figure 6.9: Main audio content classification results for each scene of the AMR WB coded audio sequence.

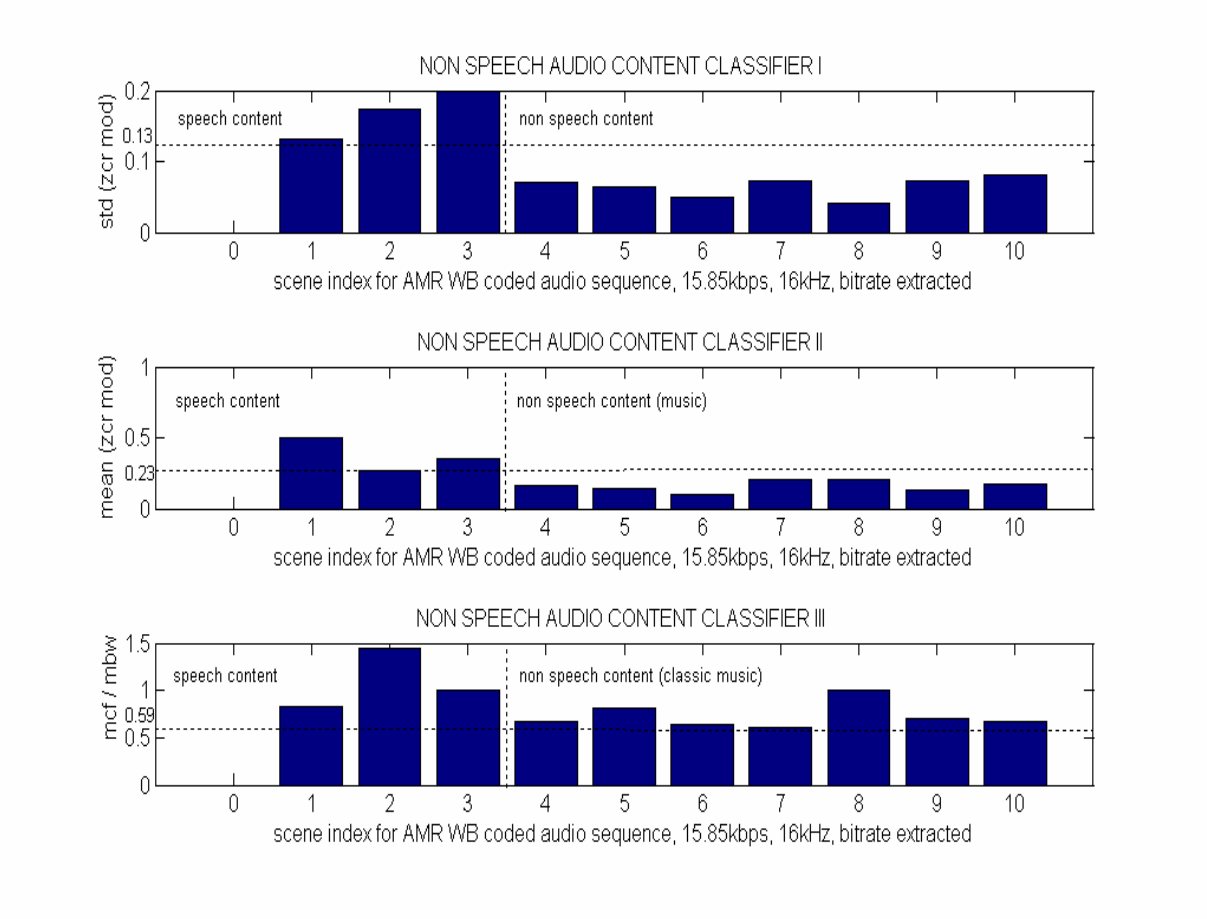


Figure 6.10: Main- and sub category audio content classification results for each scene of the AMR WB coded audio sequence.

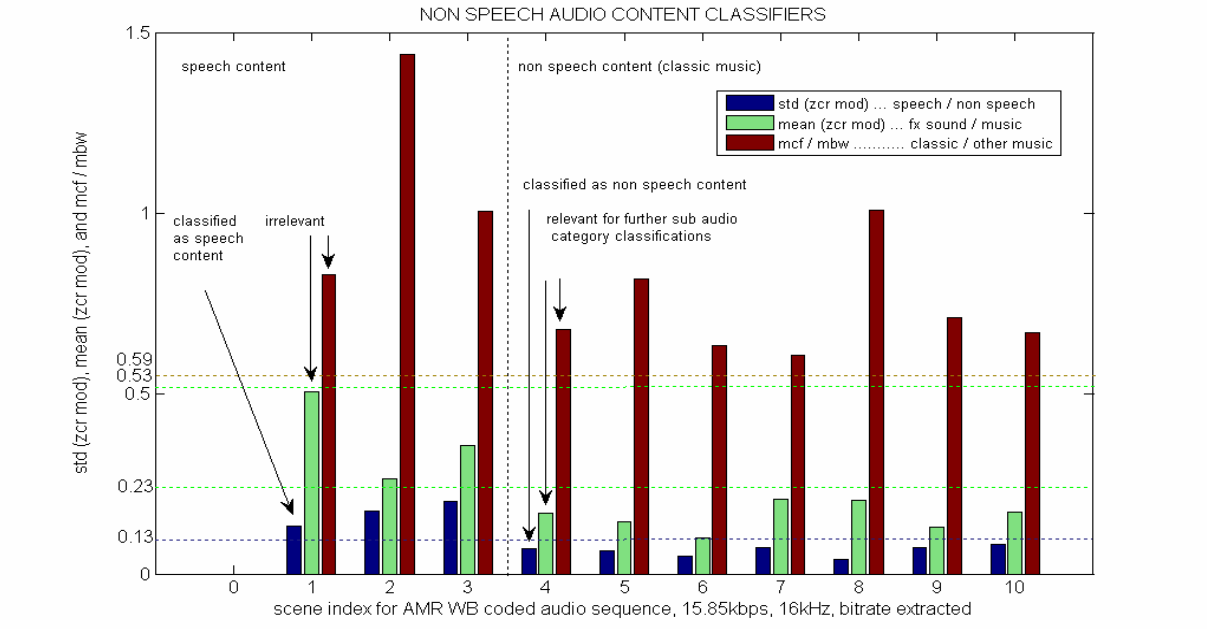


Figure 6.11: Audio content classification results for each scene of the AMR WB coded audio sequence.

Comparing the results for coded audio content classification for the case of known audio codec settings bitrates and sampling frequencies with the results from coded audio content classification for unknown audio codec settings, there are no significant differences.

Results for audio quality estimation of each scene for known audio codec settings and the correlation vector interpretation for those results are shown in Fig.6.12 and Fig.6.13:

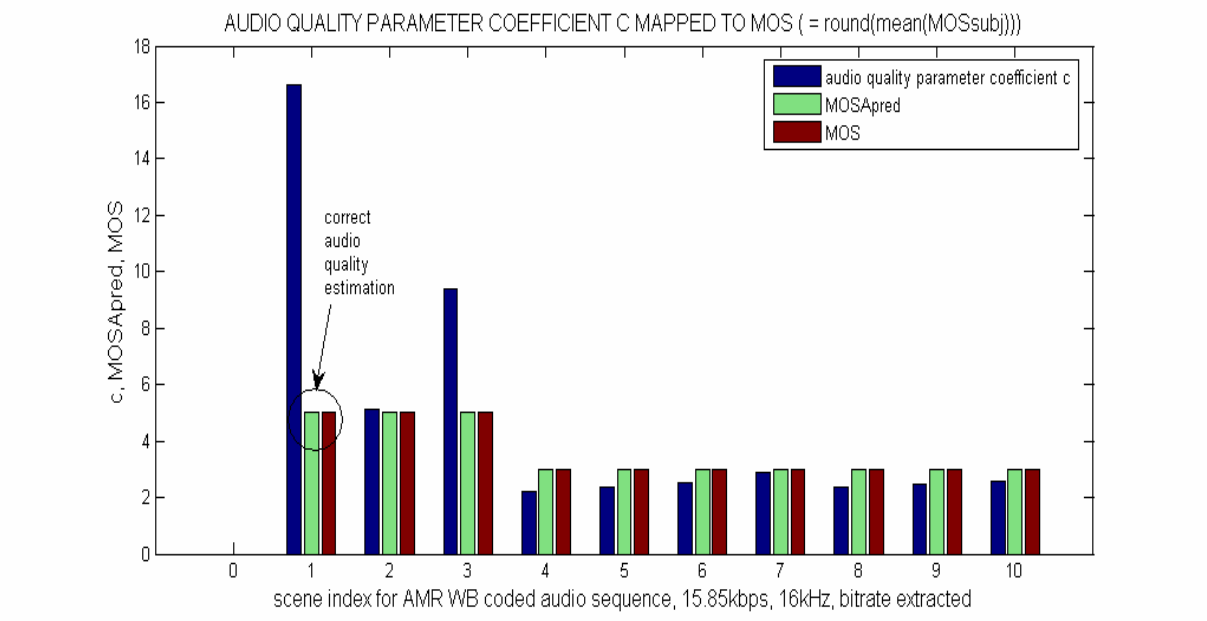


Figure 6.12: Audio quality estimation and mapping results for each scene of the AMR WB coded audio sequence with different audio content.

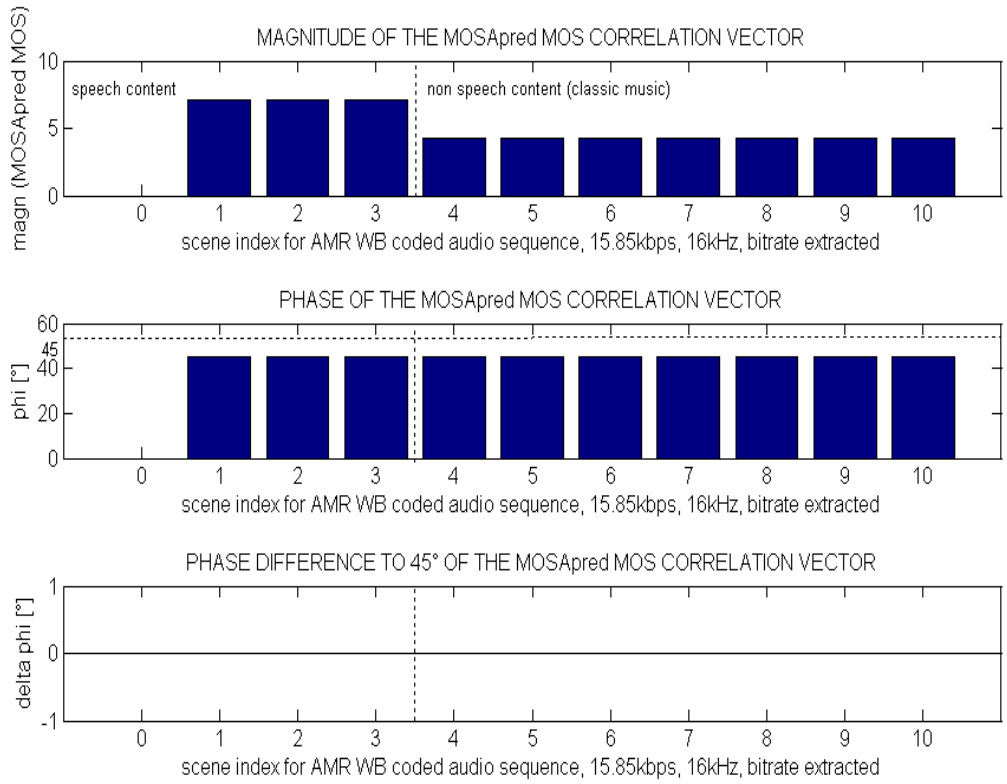


Figure 6.13: Audio quality estimation and mapping results for each scene of the audio sequence, correlation vector representation.

Comparing this results with those of unknown audio codec settings, it can be seen, that for the case of known audio codec settings the audio quality estimation for all scenes are correct, even for scene 1 and scene 3. The audio quality of the first three AMR WB coded speech scenes with15.85kbps, sampled at 16kHz is correctly estimated as “5 ... excellent”, and the audio quality of the AMR WB coded classic music scenes with 15.85kbps, sampled at 16kHz is correctly estimated as “3 ... fair”. Comparing both classification results from the audio quality estimation for unknown and known audio codec setting characteristics, it can be seen, that there are no significant differences in the classification and estimation results of each stage of the whole audio quality estimation system.

Program outputs and results for the audio quality estimation of the whole audio sequence and for one specific audio scene with known audio codec settings are given in appendix C.

Chapter 7

Conclusion

Main topic of this diploma thesis was audio quality estimation for video sequences for multimedia content in UMTS networks. Therefore, the perceived audio quality of the end user was investigated under specific point of views. The research area in this diploma thesis is focused on low bitrate mobile audio services and reference free audio quality estimation for AAC, AMR WB, and AMR NB coded different audio content. First, the perceived audio quality impression of different coded audio content was estimated by subjective MOS listener tests. Those subjective listener tests have shown, that the most suitable audio codecs and audio codec settings for the different audio contents are those, which are resumed in Table 7.1:

audio content	audio codec settings	audio quality rated as
speech	AMR WB 15.85kbps, 16kHz	excellent 5
other music	AAC 24kbps, 22.05kHz	excellent 5
classic music	AAC 20kbps, 16kHz	excellent 5
ambient, fx sounds	AMR WB 15.85kbps, 16kHz	excellent 5

Table 7.1 Most suitable audio codecs and audio codec settings for different audio content types.

With those specific audio codec, audio codec settings, and audio content information it was possible to design a reference free audio quality estimation metric or system, without using reference based objective measurement algorithms. The whole information to predict the perceived audio quality reference free is extracted in the frequency domain from the coded audio file without reference source in form of the original, uncoded audio file. While the frequency spectrum of a coded audio file includes all necessary information about the used audio codec, audio codec settings bitrate and sampling frequency, and audio content, a linear

reference free audio quality estimation metric was designed together with the audio codec and audio content specific results from the subjective MOS listener tests mean(MOS_{subj}). A linear metric parameter p, proportional to the mean value of the audio codec and audio content
specific subjective MOS listener test result, was found in the frequency domain in form of the mean value of the magnitudes over all frames of a coded audio file frequency spectrum.

To predict the perceived audio quality by a reference free linear audio quality metric, it is necessary to identify the audio codec, audio codec settings bitrate and sampling frequency, and audio content by classification algorithms. Once, all of those audio file characteristics are classified, the corresponding mean value from the subjective MOS listener tests result can be chosen from a lookout table for estimating the perceived audio quality MOS_{Apred} in the following way: first, the audio quality parameter coefficient c is calculated as the ratio of the codec and content specific mean(MOS_{subj}) and the extracted audio quality parameter p, as resumed in equation 7.1:

$$c = \text{mean}\,(MOS_{subj}) / p \tag{7.1}$$

or

$$\text{mean}\,(MOS_{subj}) = c \cdot p \tag{7.2}$$

The predicted audio quality MOS_{Apred} is then calculated by mapping the specific audio quality parameter coefficient c to the rounded version of the corresponding mean(MOS_{subj}) value, as resumed in equation 7.3:

$$MOS_{Apred} = \text{mapped}(\,c\,) = \text{round}\,(\text{mean}(MOS_{subj})) \tag{7.3}$$

or, c expressed in terms of mean(MOS_{subj}) and p:

$$MOS_{Apred} = \text{mapped}\,(\text{mean}(MOS_{subj})/p) \tag{7.4}$$

As mentioned above, audio codec, audio codec settings, and audio content classifications are necessary to identify the corresponding mean(MOS_{subj}) value for predicting the audio quality MOS_{Apred}. Therefore, classifiers were found in the time- and frequency domain, such as:

- the mean centre phase mean phase range ratio for AAC / AMR codec classification
- the standard deviation of the phase for AMR WB / AMR NB classification
- the mean centre phase for AAC bitrate and sampling frequency classification
- the standard deviation and mean value combination of the zero crossing rate for AMR WB and AMR NB bitrate and sampling frequency classification
- the standard deviation of the zero crossing rate of a modified coded audio file version for speech / non speech content classification
- the mean value of the zero crossing rate of a modified coded audio file version for fx sounds / music classification
- the mean centre frequency and mean bandwidth ratio for classic music and other music classification

All of this classifier and their threshold values are resumed in Table 7.2:

classification of	classifier	Threshold value
AAC / AMR	mcp / mpr	> 3.2 : AAC
AMR WB / AMR NB	std (phase)	> 0.19: AMR NB
AAC bitrate, sampling frequency	mcp	see Fig. 5.8*
AMR bitrate, sampling frequency	mean(zcr), std(zcr)	combination of both*
speech / non speech	std (zcr_mod)	> 0.13: speech
fx sound / music	mean (zcr_mod)	0.23 < mean(zcr_mod) < 0.53
classic music / other music	mcf / mbw	> 0.59: classic music
MOS_{Apred}	round(mean(MOS_{subj})) mapped(mean(MOS_{subj})/p) mapped(c)	see Table 5.2*

Table 7.2: Classifiers and threshold values for reference free audio codec, audio codec settings, audio content, and audio quality estimation.

*) … classification based on more than one threshold values (intervals), of classifier combinations, or codec and content specific classification / mapping intervals.

Testing the whole audio quality estimation system by using a test setup of 349 different coded audio files with different audio contents (see appendix C.2.1), a correct audio quality MOS_{Apred} prediction of 70.49% is possible with a Pearson linear factor of 0.867.

By extending the whole audio quality estimation system by an scene detection tool (cf. chapter 6), it is possible, to predict the audio codec, audio codec settings, audio content, and audio quality of each scene of a coded audio sequence, extracted from a video clip. Further, this extended version of the audio quality estimation system enables the prediction of those characteristics for the whole audio sequence.

Finaly, this reference free audio quality estimation system avoids the disadvantages of cost- and time consuming subjective mean opinion score (MOS) listener tests and reference based objective quality measurement mehods.

Appendix A

Multimedia streaming in UMTS networks

A.1 Introduction

Streaming is a method for transferring data with real-time characteristics [32], so that the user (recipient) can start viewing the presentation before the entire contents have been transmitted. A streaming platform supports a multitude of different multimedia applications and contents, at various bitrates and qualities. For example, news at very low bitrates using still images and speech, music listenings, video clips and watching live sports events. Such streaming platforms are supporting also progressive downloading of media for selective media types, which can be further protected with a standardised digital rights management (DRM) technology.

During the streaming process, a server sends multimedia content to an user (client or recipient) over a network in real-time. In a transparent end-to-end packet switched streaming service (PSS), the multimedia content is divided into packets by the streaming server to make a transmission over the network possible. Such a transparent end-to-end packet-switched streaming service PSS is specified by the third generation partnership project (3GPP) in which multimedia streaming packets are reassembled by the user at the receivers end, which enables the user to play the multimedia content as it comes in. A series of related packets is called a stream. The data packets are sent in real time and time-stamped, so they can be displayed in time-synchronized order. There are two possibilities to transmit such real time strings: unicast or multicast. In the case of unicast, the real time streams are sent from one server to one client (one-to-one), and in the case of multi cast, the real time streams are sent from one server to more than one client (one-to-many). The main difference from simple file transfer and streaming lies in the fact, that the client can play the multimedia content as it comes in over the network, rather than waiting for the entire multimedia content to download before it can be played. For streaming, the multimedia service must be able to relate media components to each other. For example, in a multimedia stream, consisting of video- and audio content, both components must be synchronized. Section 2.2-2.4 gives an overview over the architecture,

streaming mechanism, streaming codecs, file formats, streaming protocols, and network elements involved in a transparent end-to-end packet switched streaming service (PSS).

A.2 Transparent end-to-end packet switched streaming service (PSS)

Transparent end-to-end packet switched streaming service (PSS) is a specification by the third generation partnership project (3GPP), which is a collaboration agreement between several telecommunication standardization bodies [34]. PSS defines a framework for an interoperable streaming service in 3GPP mobile networks and is an application level service that mostly deals with client and server. Although streaming can benefit from network support (e.g. Quality of Service QoS), one requirement for PSS is that it should work over different (QoS) bearers. Therefore, multimedia services should be designed in such a way that they can be adapted to the network.

The basic framework appeared the first time by simple streaming services in Release 4 of the 3GPP specification. 3GPP Release 5 had introduced extended features such as capability exchange while the backward compatible 3GPP Release 6 completes the PSS feature set to a comprehensive content delivery framework. 3GPP release 6 framework updates the list of recommended media types and codecs to achive higher service quality within the 3GPP environments. An overview of network elements involved in a 3G packet switched streaming service are presented in Fig.A.1 [33]:

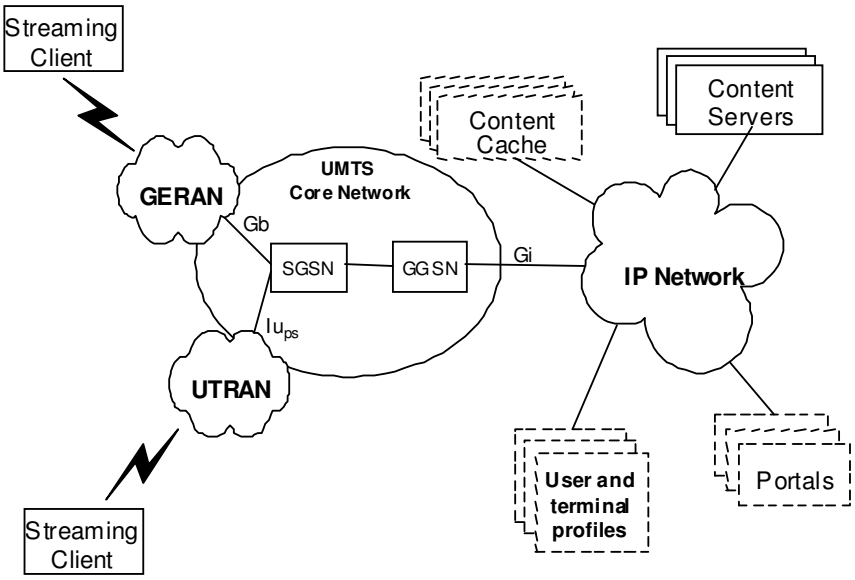


Figure A.1: Network elements involved in a 3G packet switched streaming service.

As Fig.A.1 shows, a successful streaming session or streaming establishment over UMTS network using a transparent end-to-end packet switched streaming service (PSS) requires the following network elements connected to the IP network:

- content server and content cache: from which the multimedia services and multimedia content can be requested by the user

- user and terminal profile server: to store user performances and terminal capabilities

- portals: those servers allow convenient access to streamed media content and are connected with the content server and content caches via IP network

The IP network is located by an Gi interface behind the UMTS core network, and the streaming clients are connected via the Radio Access Networks GERAN or UTRAN, and the following chapter gives an overview of the transparent end-to-end packet switched streaming service PSS over UMTS network.

A.3 Streaming scenario in UMTS

A simple streaming session in a packet switched streaming application (PSS) consists of streaming control protocols, transport protocols, media codecs, and scene description protocols. At the beginning of every streaming session, the mobile user gets an universal resource identifier (URI), specifying a streaming server and the content address on that server. There are three possibilities how a mobile user can get an URI: from a world wide web (WWW) browser, from a wireless application protocol (WAP) browser, or typed by hand. The description of a streaming session in relation to the requested content characteristics (session name, author, ...), media type, and transmission bitrate is given in the Session Description Protocol (SDP) file. This SDP file can be given in the link of a HTML page, through a Real-Time Streaming Protocol (RTSP) [35], or via a Multimedia Session Service (MMS). Here, the MMS user agent receives a modified MMS message from the MMS relay or server.

Every streaming session starts with the so called session establishment, a process, in which the browser or the mobile user invokes a streaming client to set up the session against the server. An active PDP context in accordance with [40] or other type of radio bearer that enables IP packet transmission is expected from the UE at the start of the session establishment signalling. The client may be able to ask for more information about the content and shall initiate the provisioning of a bearer with appropriate QoS for the streaming media.

The basic RTSP unicast operation first appeared in 3GPP Release 4, in which the client or mobile user gets an universal resource identifier (URI), which specifies a streaming server and the address of the content on that server, from a WWW-browser or WAP-browser, or learns the location of a media clip by browsing to a web page that has an RTSP URI. The streaming player connects to the streaming server and issues a RTSP DESCRIBE command. The server responds with an Session Description Protocol (SDP) which includes information like number of streams, media types, and required bandwidth. After parsing the description, the client issues an RTSP SETUP command for each stream in the session. The SETUP command tells the server which ports the client uses to receive the media. When streams have been set up, the client issues a RTSP PLAY command after which the server starts sending one or more media streams as RTP packets over the IP network. Finally, the client issues a

TEARDOWN command to end the streaming session. This schematic view of a streaming session is illustrated in [33].

In 3GPP Release 5, during the streaming initiation, the client provides a capability profile to the server (an URL reffering to the profle and possible differences) [33]. With the PSS capability profile, the client can send information about its available number of audio channels, supported media types, rendering screen size, and bits per pixel to the server, which further uses this information to select the most suitable content for the client.
3GPP Release 6 consists of already defined download and streaming framework appended with alternative of progressive downloading in an end-to-end delivery context. This enables optional use of strong content encryption and integrity protection capabilities and interoperability with cryptographic key management systems.

Between PSS providers a standardised container file exchange is possible as a specific server file format. For session bandwidth adaption to the potentially time-varying cellular network bandwidth, PSS allows the selection of streaming session alternatives (alternative SDP) and dynamic, link-aware bandwidth adaptation. This feature is especially useful in cellular networks where QoS-enabled bearers are not available. 3GPP Release 6 also presents a defined mechanism to gather streaming session Quality-of-Experience metrics at the PSS service provider's premises. The capability exchange mechanisms in Release 6 enable better service filtering for both streaming and static media contents. Further, a progressive downloading mechanism is implemented in Release 6 to start media playback during the media "download". The 3GPP PSS provides a framework for Internet Protocol (IP) based streaming applications in 3G networks.

A.3.1 Streaming protocols in UMTS

Streaming Protocols are essential for PSS to control session establishment, session-setup, capability exchange, session control, scene description, and data transport of streaming media and other data. The following streaming related protocols utilize TCP / IP [38] and / or UDP / IP [39] as their transport [32]:

- RTP: Real-Time Transport Protocol [RFC 1889, RFC 1890]
  Provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio or video, over multicast or unicast network services. In PSS, RTP is carried only over UDP and does not guarantee quality-of-service for real-time services,

- RTCP: Real-Time Control Protocol [RFC 1889]
  The primary function of RTCP is to provide feedback on the quality of data distribution, which is achieved by periodic "receiver report" packets sent by the receiver to the sender, containing inter-arrival jitter measured by the receiver and number of packets lost [35];

- RTSP: Real-Time Streaming Protocol [RFC 2326]
  RTSP is used to establish and control time-symchronized streams of continous media and acts as a "network remote control" for multimedia services;

- SDP: Session Description Protocol [RFC 2327, RFC 2326]
  The purpose of SDP is to convey information about media streams in multimedia sessions to allow the recipients of a session description to participate in the session,

- UDP/IP: User datagram protocol [RFC 768]
  for transport of RTP flows and data,

- TCP/IP: Transmission control protocol [RFC 793]:
  for transport of data only,

- SCTP/IP: Stream control transmission protocol [RFC 3286]
  IP media transport protocol that also provides telephony signaling transport critical functions.

For the transport of session control and media data, PSS clients and server shall support an IP-based network interface. Control and media data are sent using the TCP / IP [38] and UDP / IP [39].

Fig.A.2 gives an overview of the protocol stack [41] used in PSS and also shows a more detailed view of the packet based network interface. The functional components can be divided into control, scene description, media codecs and the transport of media and control data.

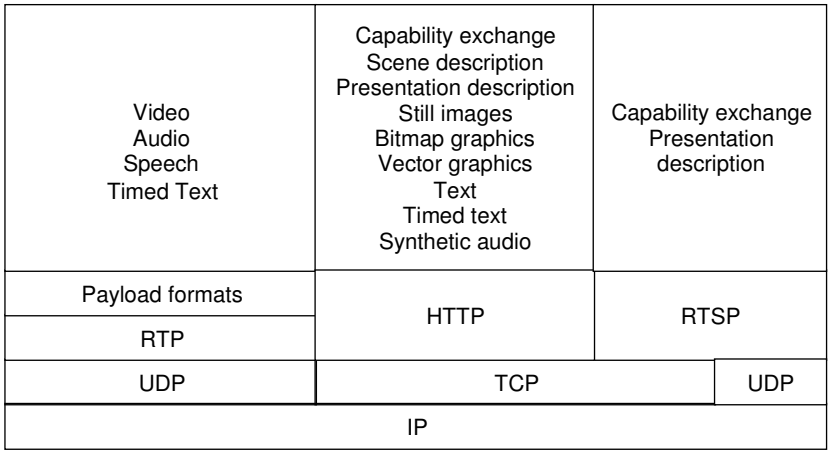


Figure A.2: Overview of the protocol stack.

A.3.2 RTP Payload formats

Information about the characteristics of each component characteristics of the multimedia content in a streaming session (video, audio, speech, time text) and the media specific RTP payload formats are seperatly specified in the Real Time Protocol (RTP) packets. The RTP payload formats are defined by the internet engineering task force (IETF) RTP [35] and [42] and provides also a protocol called Real Time Control Protocol (RTCP (see clause 6 in [40]) for feedback about the transmission quality. Further, the User Datagram Protocol and the

media IP protocol UD/IP for the transport of continous media (speech, audio and video) shall be supported and the media IP protocol, following the media specific RTP payload formats, shall be used. Those content depended, media specific RTP payload formats are:

- AMR narrow band speech codec RTP payload format according to [36]. A PSS client is not required to support multichannel sessions

- AMR wide band speech codec RTP payload format according to [36]. A PSS client is not required to support multichannel sessions

- MPEG-4AAC audio codec RTP payload format according to RFC 3016 [38]

- MPEG-4 video codec RTP payload format according to RFC 3016 [38]

- H.263 video codec RTP payload format according to RFC 2429 [39]

A.3.3 UMTS streaming codecs

Before transferring a media in realtime, which is called streaming, the media has to be encoded. This could involve compression, which might impact on the media quality (distortion) depending on the compression level. An overview of encoding and transport technologies for each media types is given in [32]. Table A.1 gives an overview of the supported UMTS streaming codecs [32].

Type	Codec (Decoder)	Support	Max. Bitrate [kbps]	Notes
Speech	AMR-NB	Required	12.2	
Speech	AMR-WB	Required	23.85	
Audio	MPEG-4 AAC-LC	Recommended	N/A	
Audio	MPEG-4 AAC-LTP	Optional	N/A	
Video	H.263 profile 0 level 10	Required	64	Max. Frame size 176x144
Video	H.263 profile 3 level 10	Recommended	64	Interactive and wireless streaming profile.
Video	MPEG-4 Simple Visual Profile Level 0	Recommended	64	Max. Frame size 176x144

Table A.1: 3GPP PSS audio and video codecs [32].

A.3.4 UMTS streaming file formats

While the streaming coding format is used to transform the multimedia content into a code stream, the specific streaming file format enables the prestored code stream for local decoding and playback, for the transfer on different media, or for different streaming transport.

UMTS streaming supports .3gp and .mp4 streaming formats [34]. The .3gp file format is defined by 3GPP [34] as a standard for multimedia streaming services over wireless networks and is based on ISO base file format [34], which allows the mixing and seperation of different media types. Table 2 gives an overview over the conformance of .3gp and .mp4 streaming file

formats to one or more several user or client profiles and examples for brands in 3GP files [34] are given in Table A.2:

Conformance	Suffix	Brand	Compatible brands	Example content
Streaming servers: Some files may in principle also be used for MMS or download.				
Release 6	.3gp	3gs6	3gs6, isom	AMR and hint track
Release 6	.3gp	3gs6	3gs6, isom	2 tracks H.263 and 2 hint tracks
Release 6, 5, 4	.3gp	3gs6	3gs6, 3gp6, 3gp5, 3gp4, isom	H.263, AMR and hint tracks
3GP file, also conforming to MP4				
Release 4, 5 and MP4	.3gp	3gp5	3gp5, 3gp4, mp42, isom	MPEG-4 video
MP4 file, also conforming to 3GP				
Release 5 and MP4	.mp4	mp42	Mp42, 3gp5, isom	MPEG-4 video and AAC

Table A.2: Conformance of different streaming file formats in UMTS [34].

A brand identifies a specification or a conformance point in a specification; its presence in a file indicates :

- that the file conforms to the specification;
- that a reader implementing that specification is given permission to read and interpret the file.

For example, the brand ‘isom’ indicates conformance to the base structure of the ISO base media file format [32]. The 3GP General profile is branded ‘3gg6’ while the 3GP Basic profile is branded ‘3gp6’ and used in MMS and PSS.

A.4 Mobile Multimedia services

3G mobile networks are characterized by their ability to carry data at much higher rates than the generations of mobile networks before (i.e., 9.6 kbps). It allows a 384 kbps packet-switched connection for downlink and 62 kbps circuit switch connection that is N-ISDN compatible. The 3G mobile networks provide significant greater bandwidth and is so suitable for new mobile services, such as enhanced multimedia applications and services A service can be classified as multimedia when it involves at least two of the following media types [43]:

- Speech: voice telecommunication (300-3400 Hz), focusing on mouth-to-ear intelligibility

- Audio: telecommunication of sound in general, focusing on fidelity. Various quality levels can be provided, high fidelity implying complete audio frequency spectrum (20-20000 Hz) and 44.1 kHz sampling.

- Video: telecommunication of full motion pictures and stills, focusing on fidelity.

- Data: telecommunication of information files (text, graphics, data), focusing on error-free transfer.

Applications, which can be built on top of streaming services, can be classified into on-demand and live information delivery applications. Examples of the first category are music and news-on-demand applications. Live delivery of radio and television programs are examples of the second category. Further, the multimedia service must be able to relate synchronized multimedia components, such as audio and video.

A.5 Quality of Service in UMTS network

The perception of an user about a delivered service, including multimedia applications and multimedia contents, can be identified with the quality of the service QoS. A number of parameters can influence this user perception, and the influencing parameters can differ as a function of the service, applications, or contents. Several studies have shown, that the main

parameters influencing the QoS perception of a transparent end-to-end packet switched streaming service (PSS) are [37]:

- Guaranteed bandwidth
- Delay
- Jitter
- Packet loss

All of those parameter influences the perceptual quality of a service and also the perceptual quality of the service content. For example, delays and packet loss during a streaming session leads to a low user quality perception. The available bandwidth enables the usage of different multimedia codecs with different codec settings to receive best possible user perception and satisfaction. Therefore, the quality of the choosen multimedia codec with its technical characteristics and codec settings within a service, can be reflected by a “mean opinion score” (MOS), assigned to the specific codec. So, the perceived QoS can be determed for the whole service, or especially for the multimedia codecs within the delivered service. A special method of specifying and grouping applications into QoS categories to indicate the importance of the different parameters as functions of the delivered service (QoS indicators) is the Class of Service CoS concept, which is defined in [37]. The end-to-end QoS concept and architecture for UMTS up to IP layer is defined in [37] and Table A.3 gives an overview of QoS classes in UMTS:

Traffic class	Conversational class conversational RT	Streaming class streaming RT	Interactive class Interactive best effort	Background Background best effort
Fundamental characteristics	Preserve time relation (variation) between information entities of the stream Conversational pattern (stringent and low delay)	Preserve time relation (variation) between information entities of the Stream	Request response pattern Preserve payload Content	Destination is not expecting the data within a certain time Preserve payload content
Example of the application	Voice	streaming video	Web browsing	background download of Emails

Table A.3: UMTS QoS classes.

For example, the QoS in case of streaming speech content can be classified or indicated by the preserve time relation or variaton between information entities of the stream or conversational pattern. The research for QoS indicators must be focused on all part of the streaming chain consisting of sender, channel and receiver. The QoS indicators for multimedia quality estimation in UMTS network should be designed on the end of the streaming chain (end user quality) and the QoS results should also be able on the sender side. At the moment, no such significant QoS indicators are defined for mobile streaming services. In ETSI, some parameters are defined from customers point of view, but they do not reflect the end user quality because the defined parameters do not fit subjective media perception. There are two different main methods to measure the perceptual quality of a service: subjective or objective quality measurement.

Appendix B

Audio coding technologies

The quality of wireless multimedia communication and services, is always limited by the given bandwidth and the choosen multimedia codec characteristics. In relation to the existing bandwidth gap between wireless and wired networks (one or two orders in their magnitude [44]), it is important to have codecs that provide better coding efficiency and coding technologies which achieved compact representations of media data over wireless networks. This is done by data compression algorithms before the multimedia content is transfered in realtime (streaming). Data compression allows the reduction of irrelevant information of the source signal and influences always the quality of the multimedia content (distortions), depending on the compression level. Current coding technologies were designed to minimize further implementation and memory costs by this way [44]. An overview of the different speech, audio and video codecs in relation to the bitrate and their applications are given in Fig.B.1:

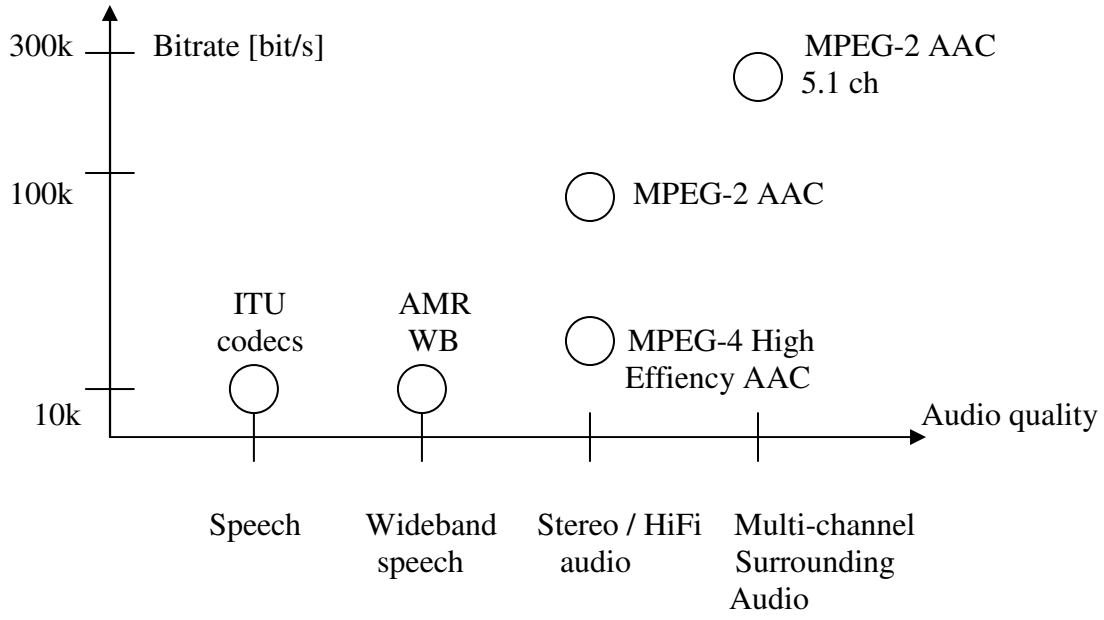


Figure B.1: Speech and audio codecs with regard to bitrate [44].

Fig.B.1 illustrates existing speech and audio coding technologies in relation to the bitrate. In 3G networks, the adaptive multirate narrowband (AMR NB, 3GPP 1999d) speech codec (encoder and decoder) is used for speech communication with 8kHz sampling rate, while adaptive multirate wideband (AMR WB, ITU-T 2002) is shown as an example of wideband speech communication, using a sample rate of 16kHz.

In comparison to media component audio, streaming media video components require a higher bandwidth. As Fig.B.2 shows, a given bandwidth of several megabits per second (Mbps) enables even the transmission video in broadcast quality video.

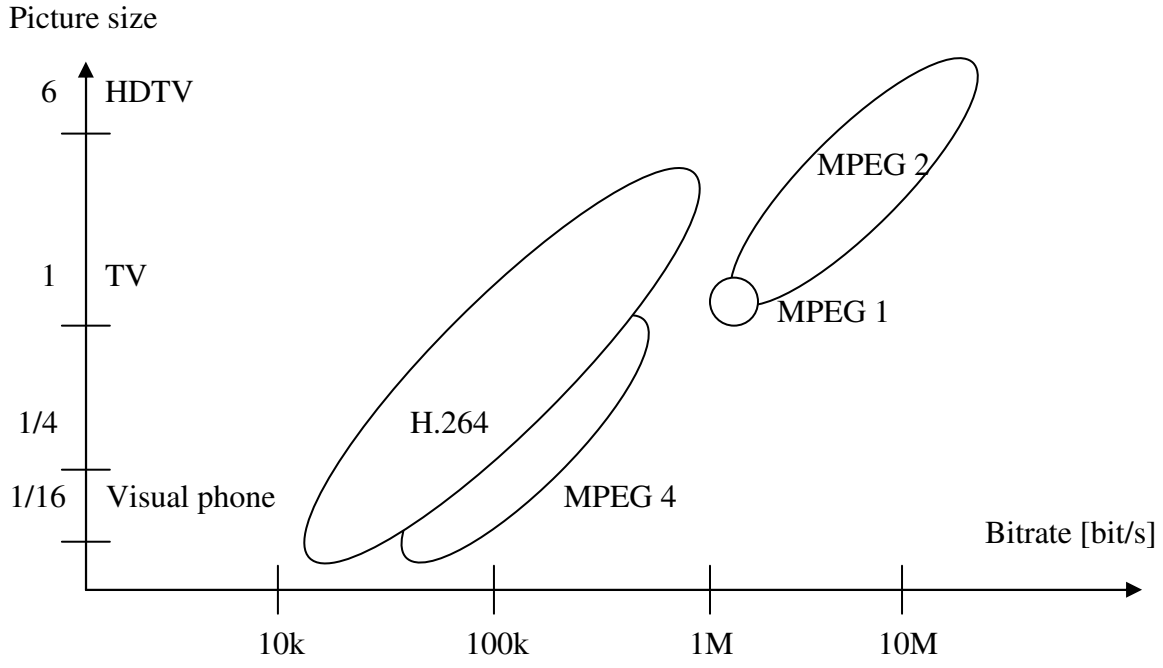


Figure B.2: Video codecs with regard to bitrate [44].

B.1 Speech and audio coding technologies

Speech and audio codecs are developed to provide a transparent perceptual reproduction of the audio information that is relevant to human auditory perception. Speech and audio codecs try to represent the speech or audio signal with a lower number of bits in comparison to the original signals for transmission or storage with best user perceived quality as possible. Main criterion for the final number of bits of the encoded speech or audio signal is the level of perceptual quality in relation to the speech or audio codec settings. The quality of a coded speech or audio signal should be close to the perceived quality of their uncoded, original versions. Bitnumber minimization or reduction is done by the coding algorithms by removing the redundand information from the original signal. Further, the quality of such digitized speech or audio signal is a function of the available audio codec settings bit rate and sampling frequeny. The availability of different bandwidths and audio codec settings, resulting in different perceptual audio qualities, leads to the development of different audio coding technologies for speech and music, based on perceptual audio quality as main design criterion. The technical characteristics of speech and audio codecs differs, for example, in the encoding / decoding delay, in the sound quality of the decoded signal, in the transmission bandwidth, and in the audio codec settings bitrates, and sampling frequencies. Further, big changes of the amplitudes in speech signals do not disturb the perceptual quality in comparison to big amplitude changes in music signals, which is relevant in relation to compression algorithm. Narrowband and wideband speech with sampling frequencies 8kHz (narrowband) and 16kHz (wideband) are handled by the most of the speech coding standards. For the speech and audio reproduction of the information that is relevant to human auditory perception, modern audio codecs employ psychoacoustic principles to model human auditory perception [44]. Such psychoacoustic principles and models are also used in the research field of Quality of Service to proof the perceptual quality of a mobile multimedia service for users. According to the main differences between speech and music, there are different principles of how speech and audio codecs are developed: speech encoders are based on a model for speech production, while for non-speech signals, like music or background noise, such a source model does not work [44].

B.1.1 Speech coding standards

Generally, speech codecs can be divided into three broad categories [44]:

1.) Waveform codecs, based on pulse code modulation (PCM, ITU-T G.711), differential PCM (DPCM), or adaptive DPCM (ADPCM, ITU-T G.726).

2.) Parametric codecs, based on linear prediction coding (LPC, FS 1015) or mixed excitation linear prediction (MELP, FS).

3.) Hybrid codecs, based on variations of the code-excited linear prediction (CELP) algorithm.

Fig.B.3 gives an overview of the speech quality in the context of coding techniques and their details are described in [44]:

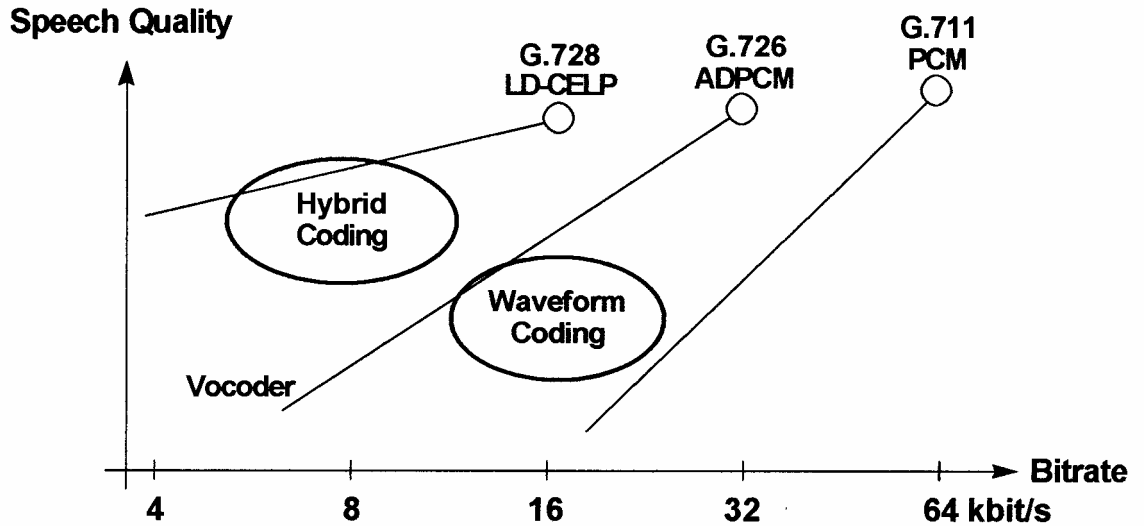


Figure B.3: Speech quality with regard to bitrate.

B.1.2 Principles of audio coding

The goal of audio coding for music and sounds is to find a compact description of the original audio signal while maintaining good perceptual quality. This means, that trained listeners (so-called golden ears) cannot distinguish the original source material from the compressed audio [44]. Further, audio codecs are developed in that way, that they reduce the effective bitrate of transmission and storage of the information data. General, there are two main audio coding principles available:

1.) audio coding based on statistic models of the signal amplitudes (linear prediction and entropy coding, "loss-less audio codecs")

2.) audio coding based on psychoacoustic models (“lossy audio codecs”)

Lossy audio compression algorithms are popular as they provide higher compression rates compared to the loss-less ones (4-12 times).While audio quality assesment and measurement methods use perceptual models to predict the audio quality of coded audio content, the following chapter gives an overview over the human perception of sound in relation to the lossy audio codecs, based on psychoacoustic models (“lossy audio codecs”).

B.1.3 The human auditory system

This section gives an overview of the sound signal processings in the human auditory system and the main psychoacoustic phenomens. Most of the modern lossy audio codecs and perceptual quality methods are developed based on psychoacoustic effects. Fig.B.4 [45] shows the main components of the human auditory system in its physiological structure which are explained in the following section:

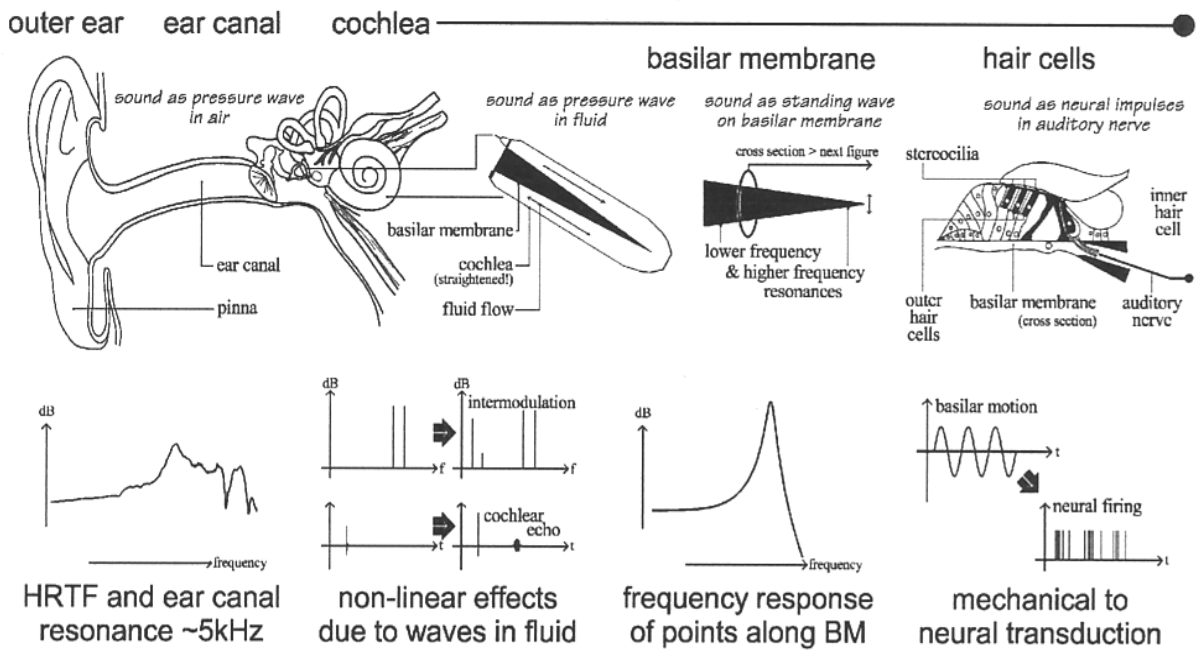


Figure B.4: Physiological structure and main components of the human auditory system [45].

The following overview gives an explanation of the human auditory main components of Fig.B.4:

Pinna:

Pre-filtering of the incoming sound (Head Related Transfer Function HRTF [46]).

Ear canal:

Filters the sound with a resonance at around 5kHz.

Cochlea:

Fluid-filled coil within the ear, partially protected by small bones.

Basilar membrane (BM):

semi-partitions the cochlea, acts as a spectrum analysator, decomposes spatially the signal into frequency components, each point on the BM resonates at a different frequency (frequency-to-place transformation), frequency selectivity is given by the width of the filter at each of this points.

Outer hair cells:
are distributed along the length of the BM, they change the resonant properties of the basilar membran by reacting to feedback from the brainstorm.

Inner hair cells:
are transforming the basilar motion to neural firing, stronger motions cause more impulses, neuronal "firing" when the basilar membran moves upwards, moment of the transformation from physical waves to physiological information [47] transducing the sound wave at each point into a signal on the auditory nerve.

Each cell needs a certain time to recover between firings, so the average response during a steady tone is lower than that at its onset. Thus, the inner hair cells act as an automatically gain control. The firing of any individual cell is pseydo-random, modulated by the movement of the basilar membrane.

Summarizing the functions of each main components in the auditory system, the whole human auditory sound signal processing system encodes an audio signal with relative wide bandwidth and large dynamic range along nerves which smaller bandwidths (narrowband) and limited dynamic ranges.The critical point is that any information lost due the transduction process within the cochlea is not available to the brain – the cochlea is effectively a lossy coder.

B.1.4 Psychoacoustic principles

Psychoacoustics deals with the relationship of physical sounds and how the human brain interpret them. The field of psychoacoustics, e.g., [15], [16], [17], [47], [48], [49], [50], [51], has made significant progress toward characterizing human auditory perception and particularly the time-frequency analysis capabilities of the inner ear. A model of the human perceptual behavior of music using psychoacoustic findings is presented in together with methods to compute the similarity between two pieces of music. An auditory model for assessing the perceived quality of coded audio signals by simulating the functionality of the human ear and its characteristics is presented in [45], where they predict the audible and

inaudible conditions in a variety of psychoacoustic listening tests. The human music perception and cognition behaviour is, e.g., explained in [17], [31]. Several psychoacoustic principles simulate the function of the human auditorial system and are used to identify the irrelevant information, which is not detectable even by well trained listeners ("golden ears"). Those psychoacoustic principles take account of the hearing thresholds, the critical band frequency analysis, the simultanous masking and the spread of masking effect along the basilar membrane and temporal masking effects.

B.1.4.1 Absolute hearing threshold

The absolute hearing threshold characterizes the amount of energy needed in a pure tone such that it can be detected by a listener in a noiseless environment [44]. Fig.B.5 shows the absolute threshold of hearing in quiet, noiseless environment. The absolute threshold is expressed in terms of Sound Pressure Level (dB SPL) [50], as equation B.1 shows:

$$L_{SPL} = 20 \log(p/p_0) \quad \text{(B.1)}$$

where

L_{SPL} … sound pressure level of a stimulus in [dB]
p ……. sound pressure level of a stimulus in Pascal (N/m²)
p_0 ….. standard reference level of 20 µPa

The curve is often referenced by audio codec designer by equating the lowest point near 4 kHz in such way that the smallest possible output signal of their decoder will be presented close to 0 dB SPL and the quiet threshold is well approximated by the non-linear function [49], [50], [51], given in equation B.2:

$$T_q(f) = 3.64(f/1000)^{-0.8} - 6.5e^{-0.6(f/1000-3.3)^2} + 10^{-3}(f/1000)^4 \quad \text{(B.2)}$$

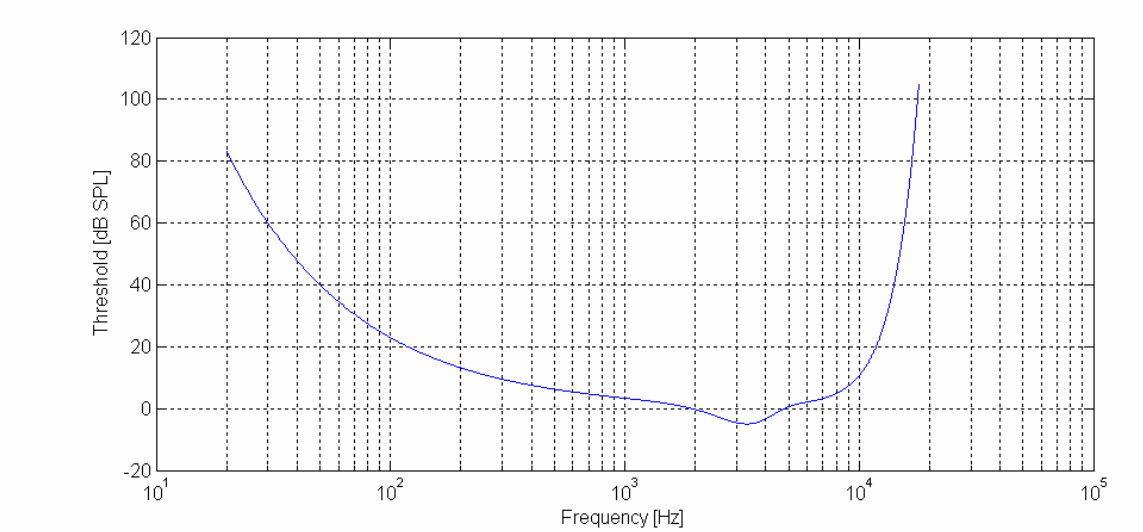


Figure B.5: Absolute hearing threshold.

B.1.4.2 Critical bands

The inner ear separates the frequencies and concentrates them at certain locations along the basilar membrane (frequency-to-place transformation), so it can be regarded as a complex system of a series of overlapping band-pass filters with asymmetrical, non-linear and level depending magnitude responses. The bandwidths of the cochlear band-pass filters are non-uniform and increases with increasing frequency. The locations (centre frequencies) and bandwidths of those cochlear band-pass filter bands in the frequency domain have been analyzed through several psychoacoustic experiments. One of the psychoacoustic models for the centre frequencies of those band-pass filters is the critical-band rate scale, where frequencies are bundled into 25 critical-bands with the unit name Bark. A distance of one critical band is commonly referred to as "one bark" and the following equation (B.3) is often used to convert from frequency in Hertz to the Bark scale [49-51]

$$z(f) = 13 \cdot \operatorname{atan}(0.76 \cdot f) + 3.5 \cdot \operatorname{atan}((f / 7.5)^2) \qquad \text{(B.3)}$$

The Bark scale is a nonlinear scale that describes the nonlinear, logarithmic sound processing in the ear and Table B.1 gives an overview of the centre frequencies and bandwidths for each of the 25 Bark bands [49-51], while Fig.B.6 shows this Hertz to Bark scale transformation

Band No.	Centre Freq. [kHz]	Bandwidth [Hz]	Band No.	Centre Freq. [kHz]	Bandwidth [Hz]	Band No.	Centre Freq. [kHz]	Bandwidth [kHz]
1	50	0 - 100	10	1.175	1.80 - 1.27	19	4.8	4.4 - 5.3
2	150	100 - 200	11	1.37	1.27 - 1.48	20	5.8	5.3 - 6.4
3	250	200 - 300	12	1.6	1.48 - 1.72	21	7.0	6.4 - 7.7
4	350	300 - 400	13	1.85	1.72 - 2.0	22	8.5	7.7 - 9.5
5	450	400 - 510	14	2.15	2.0 - 2.32	23	10.5	9.5 - 12
6	570	510 - 630	15	2.5	2.32 - 2.7	24	13.5	12 - 15.5
7	700	630 - 770	16	2.9	2.7 - 3.15	25	19.5	15.5 -
8	840	770 - 920	17	4.3	3.15 - 3.7			
9	1000	920 - 1080	18	4.0	3.7 - 4.4			

Table B.1: Centre frequencies and bandwidth for each of the 25 Bark bands.

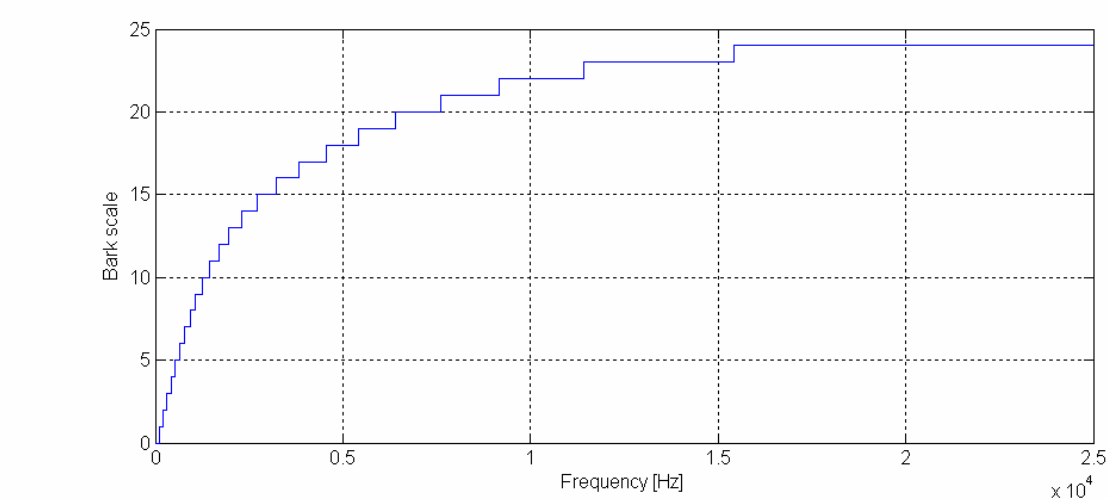


Figure B.6: Hertz to Bark Band Transformation.

The “critical bandwidth” is a function of the frequency that quantifies the cochlear band-pass filter and is conveniently approximated by equation (B.4) [49-51]

$$BWc(f) = 25+75[1+1.4(f/1000)^2]^{0.69} \quad \text{(B.4)}$$

B.1.4.3 Masking phenomens

Masking is the phenomenon where the perception of one sound is obscured by the perception of another sound and can be explained in the frequency- and time domain.

B.1.4.3.1 Frequency masking

Simultaneous masking and the spread of masking refers to a frequency-domain phenomen that can be observed whenever two or more stimuli are simultaneously presented to the auditory system. The response of the auditory system is nonlinear and the perception of a given tone is affected by the presence of other tones. The auditory channels for different tones interfere with each other, giving rise to a complex auditory response called frequency masking. That means, that a single (masking) tone is surrounded by its so-called masking threshold curve and masking bandwidth, which intersect the threshold of the hearing curve at two points. Every single tone within this masking bandwidth with its sound pressure level SPL falling below the masking curve will be overshadowed (or "masked") and will not be audible. The masking bandwidth depends on the frequency of the masking tone and increases with the SPL-value of the masking tone. Thus, louder tones with higher frequencies will mask more neighbouring frequencies than softer tones with lower frequencies. So, ignoring the frequency components in the masking band whose levels fall below the masking curve does not cause any perceptual loss. Masking phenomens are explained and shown in [44], [48], [46], [50-51].

B.1.4.4 Temporal masking

Temporal masking explains the masking effect of a softer test tone by the presence of a stronger one (mask tone) in the time-domain. The level of the masked signal depends on the time between the masker and the test tone. The stronger tone will mask softer tones with

lower levels, which appears a short time later (decaying time). This temporal masking effect based on the functionality of the human auditory system, that the inner hair cells within the human ear needs a "recovery time" from the strong mask tone, until they are able to realize the existence of the softer test tone. In the case of audio signals, abrupt signal transients create pre- and post- masking regions in time during which a listener will not perceive signals beneath the audibility thresholds produced by the masker. Pre- or “backward masking” is based on processing times in the ear and means, that signals just before the strong masker appears are masked [46].

B.1.5 Audio Codecs based on psychoacoustic models

Audio codecs based on psychoacoustic models tries to take advantage of the way the human auditory system perceives sound. Those lossy or loss less audio codecs take advantage of the perceptual characteristics of the human auditory system like absolute hearing threshold, simultaneous masking, spread of masking along the basilar membrane and temporal masking, as descriped above. Those psychoacoustic phenomens are implemented in the audio codec encoder side in form of a psychoacoustic model, controlling the quantization, encoding, and reduction process of the redundant bits for the final outgoing bitstream. Comparing with speech coding technologies the general problem in audio coding is, that no unified source model for audio signal production exists, as for the case of speech signals.

B.1.5.1 Audio coding standards

As mentioned above, audio codec design is influenced by the resulting perceptual coding quality, application constraints (one-way vs. two-way communications, playback, streaming, etc.), signal characteristics, implementation complexity, and resiliency to communication errors [44]. International standards for high-quality and high-compression perceptual audio coding has been produced by the Moving Pictures Experts Group (MPEG), and the activities of this standardization body have been culminated in a number of successful and popular coding standards. In 1992, the MPEG-1 audio standard was completed and in 1994, its backward-compatible extension MPEG-2 BC was finalized. A further, more efficient audio coding standard is MPEG-2AAC, while the audio codec standard MPEG-4 was issued in 1999. These standards support audio encoding for a wide range of data rates and are used in

many applications. Table B.2 gives an overview of the MPEG audio coding standards, and their details are described in [44], [51]:

Standard	Sampling rates	Bitrates	Applications
MPEG-1	32, 44.1, 48 kHz	32-320 kbps	Broadcasting, storage, multimedia, telecommunication
MPEG-2 BC	16, 22.05, 24, 32, 44.1, 48 kHz	64 kbps/channel	Multichannel audio
MPEG-2 AAC	16, 22.05, 24, 32, 44.1, 48 kHz	48 kbps/channel	Digital television and high-quality audio
MPEG-4 AAC	8-48 kHz	24-64 kbps/channel	High quality, lower latency

Table B.2: Overview over MPEG audio coding standards [44], [51].

The audio compression used in MPEG is used by itself for music recording (MP3). The requirement for fidelity is that high frequencies must be well enough preserved so that overtone sequences allow the listeners to distinguish the different kinds of music instruments. Those high frequency preservation corresponds directly with the perceived audio quality of listeners and also with the audio codec settings bitrate and sampling frequency. High bitrates and high sampling frequencies preserve those frequency areas much more better than low audio codec settings, resulting in a better perceived audio quality. The influence of different audio codec settings on those frequency areas are shown in the Fig.5.22 of section 5.4, and can be used to predict the perceived audio quality of coded audio content, together with the results of subjective MOS listener test.

Appendix C

Matlab implementation of the reference free audio quality estimation system

C.1 Overview

The whole reference free audio quality estimation system, realized by MATLAB files or functions, can be applied for predicting the perceived audio quality of

- each audio file from an audio file list (cf. appendix C.2), consisting of different audio codecs, codec settings and contents (audio_quality.m)

- for a specific single audio file (audio_quality_single_audiofile.m)

- a randomly generated coded audio file, where audio codec, audio codec settings, and audio content are randomly generated (aq_rcc.m)

- an audio sequence and each audio scene extracted from a video clip (audio_video.m).

For all those MATLAB programs, the data source, on which the coded audio .wav and video .avi files are stored, can be chosen individually.

C.2 Different coded audio content test file setup

For testing the whole audio quality estimation system, the following 349 different coded audio files with different audio content were used (all_cod_cont.txt):

```
aac\8000\speech_1_teledoc.wav
aac\8000\speech_2_teledoc.wav
aac\8000\speech_angel.wav
aac\8000\speech_astrology.wav
aac\8000\speech_bayern.wav
aac\8000\speech_cnn_1.wav
aac\8000\speech_cnn_2.wav
aac\8000\speech_cnn_3.wav
aac\8000\speech_euronews_1.wav
aac\8000\speech_eurosport_1.wav
aac\8000\speech_eurosport_2.wav
aac\8000\speech_HSE24.wav
aac\8000\speech_kika_1.wav
aac\8000\speech_kika_2.wav
aac\8000\speech_matrix_1.wav
aac\8000\speech_matrix_2.wav
aac\8000\speech_roy.wav
aac\8000\speech_sat1.wav
aac\8000\speech_stadt.wav
aac\8000\speech_zib_1.wav
aac\8000\speech_zib_2.wav
aac\8000\speech_zib_boerse.wav
aac\16000\speech_1_teledoc.wav
aac\16000\speech_2_teledoc.wav
aac\16000\speech_angel.wav
aac\16000\speech_astrology.wav
aac\16000\speech_bayern.wav
aac\16000\speech_cnn_1.wav
aac\16000\speech_cnn_2.wav
aac\16000\speech_cnn_3.wav
aac\16000\speech_euronews_1.wav
aac\16000\speech_eurosport_1.wav
aac\16000\speech_eurosport_2.wav
aac\16000\speech_HSE24.wav
aac\16000\speech_kika_1.wav
aac\16000\speech_kika_2.wav
aac\16000\speech_matrix_1.wav
aac\16000\speech_matrix_2.wav
aac\16000\speech_roy.wav
aac\16000\speech_sat1.wav
aac\16000\speech_stadt.wav
aac\16000\speech_zib_1.wav
aac\16000\speech_zib_2.wav
aac\16000\speech_zib_boerse.wav
aac\20000\speech_1_teledoc.wav
aac\20000\speech_2_teledoc.wav
aac\20000\speech_angel.wav
aac\20000\speech_astrology.wav
aac\20000\speech_bayern.wav
aac\20000\speech_cnn_1.wav
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```
aac\20000\speech_cnn_2.wav
aac\20000\speech_cnn_3.wav
aac\20000\speech_euronews_1.wav
aac\20000\speech_eurosport_1.wav
aac\20000\speech_eurosport_2.wav
aac\20000\speech_kika_1.wav
aac\20000\speech_kika_2.wav
aac\20000\speech_matrix_1.wav
aac\20000\speech_matrix_2.wav
aac\20000\speech_roy.wav
aac\20000\speech_sat1.wav
aac\20000\speech_stadt.wav
aac\20000\speech_zib_1.wav
aac\20000\speech_zib_2.wav
aac\20000\speech_zib_boerse.wav
aac\24000\speech_1_teledoc.wav
aac\24000\speech_2_teledoc.wav
aac\24000\speech_angel.wav
aac\24000\speech_astrology.wav
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aac\24000\speech_cnn_3.wav
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aac\24000\speech_eurosport_1.wav
aac\24000\speech_eurosport_2.wav
aac\24000\speech_kika_1.wav
aac\24000\speech_kika_2.wav
aac\24000\speech_matrix_1.wav
aac\24000\speech_matrix_2.wav
aac\24000\speech_roy.wav
aac\24000\speech_sat1.wav
aac\24000\speech_stadt.wav
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amr_wb\6600\speech_matrix_2.wav
amr_wb\6600\speech_roy.wav
amr_wb\6600\speech_sat1.wav
amr_wb\6600\speech_stadt.wav
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amr_wb\8850\speech_astrology.wav
amr_wb\8850\speech_bayern.wav
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amr_wb\8850\speech_eurosport_1.wav
amr_wb\8850\speech_eurosport_2.wav
amr_wb\8850\speech_kika_1.wav
amr_wb\8850\speech_kika_2.wav
amr_wb\8850\speech_matrix_1.wav
amr_wb\8850\speech_matrix_2.wav
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amr_wb\8850\speech_sat1.wav
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amr_nb\7950\speech_stadt.wav
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aac\8000\classic_music_panorama
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aac\8000\classic_music_tw1
aac\8000\classic_music_zib_1
aac\8000\classic_music_zib_2
aac\16000\classic_music_haydn
aac\16000\classic_music_orf_1
aac\16000\classic_music_panorama
aac\16000\classic_music_teledoc
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aac\16000\classic_music_zib_1
aac\16000\classic_music_zib_2
aac\20000\classic_music_haydn
aac\20000\classic_music_orf_1
aac\20000\classic_music_panorama
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aac\20000\classic_music_zib_2
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amr_wb\8850\classic_music_zib_2
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amr_wb\12650\classic_music_orf_1
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amr_wb\15850\classic_music_zib_1
amr_wb\15850\classic_music_zib_2
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aac\8000\other_music_f4
aac\8000\other_music_matrix_1
aac\8000\other_music_matrix_2
aac\8000\other_music_roy
aac\16000\other_music_FA
aac\16000\other_music_f4
aac\16000\other_music_matrix_1
aac\16000\other_music_matrix_2
aac\16000\other_music_roy
aac\20000\other_music_FA
aac\20000\other_music_f4
aac\20000\other_music_matrix_1
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aac\20000\other_music_matrix_2
aac\20000\other_music_roy
aac\24000\other_music_FA
aac\24000\other_music_f4
aac\24000\other_music_matrix_1
aac\24000\other_music_matrix_2
aac\24000\other_music_roy
amr_wb\6600\other_music_FA
amr_wb\6600\other_music_f4
amr_wb\6600\other_music_matrix_2
amr_wb\6600\other_music_roy
mr_wb\8850\other_music_FA
amr_wb\8850\other_music_f4
amr_wb\8850\other_music_matrix_2
amr_wb\8850\other_music_roy
amr_wb\12650\other_music_FA
amr_wb\12650\other_music_f4
amr_wb\12650\other_music_matrix_2
amr_wb\12650\other_music_roy
amr_wb\15850\other_music_FA
amr_wb\15850\other_music_f4
amr_wb\15850\other_music_matrix_2
amr_wb\15850\other_music_roy
aac\8000\non_speech_stadion_euronews_1.wav
aac\8000\non_speech_stadion_eurosport_1.wav
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aac\8000\non_speech_fx_matrix.wav
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aac\16000\non_speech_stadion_eurosport_1.wav
aac\16000\non_speech_fx_kika_1.wav
aac\16000\non_speech_fx_matrix.wav
aac\20000\non_speech_stadion_euronews_1.wav
aac\20000\non_speech_stadion_eurosport_1.wav
aac\20000\non_speech_fx_kika_1.wav
aac\20000\non_speech_fx_matrix.wav
aac\24000\non_speech_stadion_euronews_1.wav
aac\24000\non_speech_stadion_eurosport_1.wav
aac\24000\non_speech_fx_kika_1.wav
aac\24000\non_speech_fx_matrix.wav
amr_wb\6600\non_speech_stadion_euronews_1.wav
amr_wb\6600\non_speech_stadion_eurosport_1.wav
amr_wb\6600\non_speech_fx_kika_1.wav
amr_wb\8850\non_speech_stadion_euronews_1.wav
amr_wb\8850\non_speech_stadion_eurosport_1.wav
amr_wb\8850\non_speech_fx_kika_1.wav
amr_wb\12650\non_speech_stadion_euronews_1.wav
amr_wb\12650\non_speech_stadion_eurosport_1.wav
amr_wb\12650\non_speech_fx_kika_1.wav
amr_wb\15850\non_speech_stadion_euronews_1.wav
amr_wb\15850\non_speech_stadion_eurosport_1.wav
amr_wb\15850\non_speech_fx_kika_1.wav
```

Single files from this list were also chosen to test audio_quality_single_audiofile.m

C.3 audio_quality.m

This MATLAB main program realizes the reference free prediction of the audio codec, audio codec settings, bitrate, sampling frequency, audio content, and audio quality of each audio file from a file list, for unknown (‘rb_est’) or known audio codec settings (‘rb_extr’). The coded .wav files and the audio lists are stored in different folders at the same location (‘C:\audio\’, ‘C:\audio_quality_audio_list’), and the audio list must be imported to the workspace. The following syntax gives an example of the audio .wav file path and name,

C:\audio\amr_nb\4750\speech_stadt.wav

and the MATLAB file audio_quality.m is called from the command line by

audio_quality(selected_cod_cont_F,
'selected_cod_cont_F_txt','selected_cod_cont_F_xls','rb_est');

if the .wav files and audio file list are stored at location F:\ and the audio codec settings are unknown. Program results are written to the .txt and .xls files 'selected_cod_cont_F_txt', and 'selected_cod_cont_F_xls', and program outputs for each audio file from the list are similar to those, which are presented in appendix C.4.

C.4 audio_quality_single_audio_file.m

This MATLAB program predicts the audio codec, audio codec settings, audio content, and audio quality of one single audio file with known or unknown audio codec settings. Path and name of the audio .wav file are similar to those described in appendix C.3, and the MATLAB file is called from the command line by

```
audio_quality_single_audiofile('F:\audio\amr_nb\4750\speech_stadt.wav',
'speech_4750_txt', 'speech_4750_xls', 'rb_est');
```

for the case of unknown audio codec settings. Detail results of the reference free audio quality estimation for one specific coded audio file are given in the following program output:

```
-------------------------------------------------------------------------------------------------------------
-- AUDIO CODEC, CONTENT, BITRATE, SAMPLING FREQUENCY AND AUDIO QUALITY -
-- ESTIMATION / CLASSIFICATION PROCESS OF .wav FILE                    --
-- C: \aac\8000\speech_stadt.wav                                       --
-------------------------------------------------------------------------------------------------------------

ZERO CROSSING RATE STATISTICS FOR CONTENT CLASSIFICATION (SPEECH / NON SPEECH)
zcr_std_modified_audiofile                          : 0.17746
zcr_mean_modified_audiofile                         : 0.25215
ratio_zcr_std_mean_modified_audiofile               : 0.70379

zcr_std_orig_audiofile                              : 0.064761
zcr_mean_orig_audiofile                             : 0.1129
ratio_zcr_std_mean_orig_audiofile                   : 0.57362

FREQUENCY DOMAIN STATISTIC FOR CONTENT CLASSIFICATION (CLASSIC/OTHER MUSIC, FX
SOUNDS)
mcf_orig                                            : 355.1897
mbw_orig                                            : 318.9002
ratio_mcf_mbw_orig                                  : 1.1138

PHASE STATISTICS I FOR AMR WB / AMR NB AND SAMPLING FREQUENCY CLASSIFICATION
std_phase_rad_orig                                  : 0.20716
mean_phase_rad_orig                                 : 0.0068241
ratio_phase_std_mean_orig                           : 30.3567

PHASE STATISTIC  II
mcp_orig                                            : 2975.0453
mpw_orig                                            : 781.4233
ratio_mcp_mpw_orig                                  : 3.8072

C:aac\8000\speech_stadt.wav
```

TIME DOMAIN STATISTIC AND RESULTS FOR AMR WB / AMR NB BITRATE AND SAMPLING FREQUENCY ESTIMATION

zcr_std_rb_orig	: 0
zcr_mean_rb_orig	: 0
Rb_est	: 8000
Rb_est_str	: 8000 bit/s
codec_id_phase_str_amr_wb_nb	: AAC
Fs_est	: 8000
Fs_est_str	: 8kHz

C:\aac\8000\speech_stadt.wav

CLASSIFICATION RESULTS FROM ESTIMATION AND BITRATE RECONSTRUCTION PROCESS :

AUDIO CODEC CLASSIFIED AS	: AAC
AUDIO CODEC BITRATE RECONSTRUCTION RESULT	: 8000
AUDIO CODEC SAMPLING FREQUENCY ESTIMATION RESULT	: 8kHz
CODED AUDIO CONTENT CLASSIFIED AS	: SPEECH CONTENT

CORRECT AUDIO CONTENT CLASSIFICATION (SPEECH) OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CODEC IDENTIFICATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CODEC BITRATE ESTIMATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CODEC SAMPLING FREQUENCY ESTIMATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CONTENT AND AUDIO CODEC CLASSIFICATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CODEC AND AUDIO CODEC BITRATE CLASSIFICATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CONTENT, AUDIO CODEC AND BITRATE CLASSIFICATION OF AUDIOFILE
C:\aac\8000\speech_stadt.wav
CORRECT AUDIO CONTENT, AUDIO CODEC, BITRATE AND SAMPLING FREQUENCY CLASSIFICATION OF AUDIOFILE

C:\aac\8000\speech_stadt.wav
CREATION OF THE .TXT FILES FOR CORRECT CLASSIFICATION SUCCESSFULL

AUDIO CONTENT CLASSIFIER FOR CODED SPEECH CONTENT
zcr_std_mod : 0.17746
AUDIO CONTENT CLASSIFIER FOR CODED NON SPEECH CONTENT MUSIC, FX SOUNDS
zcr_std_mod, zcr_mean_mod, mcf/mbw_orig : 0.17746
0.25215
1.1138
AUDIO CODEC AAC / AMR CLASSIFIER mcp_orig / mpr_orig : 3.8072
AUDIO CODEC AMR WB / AMR NB CLASSIFIER std_phase_rad_orig : 0.20716
AUDIO CODEC AMR WB / AMR NB BITRATE AND SAMPLING FREQUENCY CLASSIFIER
zcr_std_rb_orig, zcr_mean_rb_orig : 0 0
AUDIO CODEC AAC BITRATE AND SAMPLING FREQUENCY CLASSIFIER mcp_orig : 2975.0453
AUDIO QUALITY PARAMETER p : 1.1851
AUDIO QUALITY PARAMETER COEFFICIENT c : 0.96366
MAPPED TO MOS SCALE VALUE : 1 ... BAD
MEAN(MOS_SUBJ) : 1.142
MAPPED TO MOS SCALE VALUE : 1 ... BAD

CODED AUDIO FILE
C:aac\8000\speech_stadt.wav
CLASSIFIED AS :
AAC 8000 CODED SPEECH CONTENT, SAMPLING FREQUENCY 8kHz
WITH AUDIO QUALITY PARAMETER C = 0.96366, MAPPED TO THE MOS SCALE VALUE : 1 ... BAD

AUDIO FILE :
C:aac\8000\speech_stadt.wav
AUDIO QUALITY PARAMETER COEFFICIENT C : 0.96366
MOSApred : 1 ... BAD
MOS (ORIGINAL ROUNDED MEAN VALUE OF MOSsubj
FROM SUBJECTIVE AUDIO LISTENER TESTS) : 1 ... BAD
RESULT FROM BITRATE RECONSTRUCTION / RECOVERY PROCESS : 8000 bit/s

AUDIO FILE :

C:aac\8000\speech_stadt.wav

MAGNITUDE, PHASE AND PHASE DIFFERENCE TO 45° OF C AND MOS : 1.49425 49.8413 4.84127
MAGNITUDE, PHASE AND PHASE DIFFERENCE TO 45° OF C AND MOSApred : 1.38875 46.0603 1.06034
MAGNITUDE, PHASE AND PHASE DIFFERENCE TO 45° OF C AND MOS : 1.38875 46.0603 1.06034

CORRECT AUDIO QUALITY ESTIMATION / CLASSIFICATION

MAGNITUDE, PHASE AND PHASE DIFFERENCE TO 45° OF MOSApred AND MOS value : 1.41421 45 0

CORRECT AUDIO QUALITY ESTIMATION / CLASSIFICATION

PEARSON LINEAR CORRELATION FACTOR : NaN
SIGNIFICANT CORRELATION < 0.05 : NaN
95 % CONFIDENCE INTERVALL, LOWER BOUND: NaN
95 % CONFIDENCE INTERVALL, UPPER BOUND: NaN

TIME ANALYSIS OF CODED AUDIOFILE:
C:aac\8000\speech_stadt.wav

FEATURE EXTRACION PROCESSING TIME FOR AUDIO CODEC AND CONTENT CLASSIFICATION:
0.118 seconds
AUDIO, BITRATE AND SAMPLING FREQUENCY ESTIMATION PROCESSING TIME:
0.0046048 seconds
CONTENT DEPENDEND AUDIO QUALITY PARAMETER ESTIMATION PROCESSING TIME :
0.1136 seconds
MOSApred PROCESSING MAPPING TIME:
0.097283 seconds

PEARSON LINEAR CORRELATION FACTOR : 1
SIGNIFICANT CORRELATION < 0.05 : NaN
95 % CONFIDENCE INTERVALL, LOWER BOUND: NaN
95 % CONFIDENCE INTERVALL, UPPER BOUND: NaN

C.4 aq_rcc.m

This MATLAB program creates a randomly coded audio file with randomly chosen audio codec, audio codec settings and audio content and predicts the perceived audio quality of such an audio file, and the MATLAB file is called from the command line by

```
aq_rcc('F');
```

where the location, on which the audio .wav files are stored, is given by the input parameter.

C.5 audio_video.m

This MATLAB program extends the whole reference free audio quality estimation system for the prediction of the audio codecs, audio codec settings, audio contents, and audio qualities of each scene in an audio sequence extracted from a video clip, for known or unknown audio codec settings. Further, the overall audio codec, audio codec settings, audio content, and audio quality are predictably. All .avi video files are stored at location ‘datasource:\video\’, and all .wav audio files are stored at location ‘datasource:\audio\’. The MATLAB file is called fom the command line by

audio_video('angel_clip.avi','amr_nb\7950\angel_clip.wav', 1024, 'rb_est','F');

using a 1024 FFT.

Appendix D

Test results of uncoded audio content classification, based on subband energy and zero crossing rate estimation, test results

Table D.1 shows the test results for uncoded audio content classification using subband energy ratio estimator and zero-crossing rate estimator, as described in chapter 3:

soundfile name	source, style	ser-value	ser-class	zcr-value	zcr-class
music_beeth_elise_1	CD, classic	0.9702	music	0.0122	music
music_beeth_elise_3	CD, classic	0.9697	music	0.0088	music
music_beeth_e_21	CD, classic	0.9707	music	0.0093	music
music_beeth_e_34	CD, classic	0.9690	music	0.0093	music
music_beeth_elise_4	CD, classic	0.9676	music	0.0086	music
music_beeth_intro_7	CD, classic	0.9363	music	0.0172	music
music_beeth_moon_1	CD, classic	0.9703	music	0.0075	music
music_beeth_moon_2	CD, classic	0.9697	music	0.0077	music
music_beeth_moon_3	CD, classic	0.9702	music	0.0074	music
music_beeth_moon_4	CD, classic	0.9685	music	0.0110	music
music_bond_1	CD, rock	0.9238	speech	0.0461	music
music_bond_2	CD, rock	0.8560	speech	0.0351	music
music_bond_3	CD, rock	0.8805	speech	0.0310	music
music_haydn_1	CD, classic	0.9419	music	0.0151	music
music_haydn_2	CD, classic	0.9059	speech	0.0195	music
music_haydn_3	CD, classic	0.9500	music	0.0140	music
music_PF_7	CD, funk	0.9044	speech	0.0602	music
music_rw_feel	CD, pop	0.9597	music	0.0229	music
music_shaolin_1	CD, wave	0.9564	music	0.0157	music
music_shaolin_2	CD, wave	0.9585	music	0.0143	music
music_shaolin_3	CD, wave	0.9582	music	0.0159	music
music_smdlvt_1	Harmonica	0.6275	speech	0.0615	music
music_smdlvt_2	CD, rock	NaN	music	0.0412	music
music_smdlvt_3	CD, rock	0.5133	speech	0.0247	music
music_wg	CD, pop	0.8270	speech	0.0476	music
music_wg_2	CD, pop	0.8753	speech	0.0358	music

music_wg_10ms	CD, pop	0.9278	speech	0.0214	music
music_wg_10ms2	CD, pop	0.6667	speech	0.0432	music
music_ziria_summer	CD, techno	0.8199	speech	0.0541	music
music_ziria_5ms	CD, techno	0.8817	speech	0.0064	music
music_ziria_100ms	CD, techno	0.7348	speech	0.0619	music
music_cm_piano	CD, classic	0.9687	music	0.0109	music
music_frozen_1	CD, wave	0.9446	music	0.0159	music
music_frozen_2	CD, wave	0.9437	music	0.0160	music
music_psb_sin	CD, pop	0.7634	speech	0.0414	music
music_psb_rent	CD, pop	0.9209	speech	0.0329	music
music_phenomena	CD, electro	NaN	music	0.0114	music
music_voc_lml_7	CD, dance	0.8445	speech	0.0551	music
music_Tspiel_1	V, ethno	NaN	music	0.0109	music
music_Tspiel_2	V, ethno	NaN	music	0.0132	music
music_Tspiel_3	V, ethno	0.8851	speech	0.0248	music
music_Tspiel_4	V, ethno	0.8382	speech	0.0232	music
music_Tspiel_5	V, ethno	NaN	music	0.0309	music
music_Tspiel_6	V, ethno	0.7396	speech	0.0300	music
music_Tspiel_7	V, ethno	0.6602	speech	0.0352	music
music_Tspiel_8	V, ethno	0.9064	speech	0.0242	music
music_Tspiel_9	V, ethno	0.8498	speech	0.0261	music
music_Tspiel_10	V, ethno	0.9272	speech	0.0211	music
music_Tmillion_1	V, electro	0.8369	speech	0.0331	music
music_Tmillion_2	V, electro	0.9454	speech	0.0256	music
music_Tmillion_3	V, electro	NaN	music	0.0161	music
music_Tmillion_4	V, rock	0.9699	music	0.0057	music
music_Tmillion_5	V, fx	NaN	music	0.0172	music
music_Tmillion_6	V, fx	0.9322	music	0.0070	music
music_Tmillion_7	V, beat	NaN	music	0.0158	music
music_Tmillion_8	V, beat	0.9570	music	0.0188	music
music_Tmillion_9	V, fx	0.8971	speech	0.0136	music
music_Tmillion_10	V, fx	0.9291	speech	0.0261	music
music_Tmillion_11	V, fx	0.8535	speech	0.0350	music
music_Tmillion_12	V, rock	NaN	music	0.0134	music
music_Tmillion_13	V, bono U2	0.9459	music	0.0192	music
music_Tmillion_1	V, bono U2	0.9129	speech	0.0185	music
music_Tmillion_15	V, bono U2	0.9051	speech	0.0179	music
music_Tmillion_16	V, bono U2	0.9155	speech	0.0190	music
music_Tjurney_1	V, ethno	0.9356	music	0.0145	music
music_Tjurney_2	V, ethno	0.9390	music	0.0142	music

music_Tjurney_3	V, ethno	0.4667	speech	0.0684	music
music_Tjurney_4	V, ethno	0.9026	speech	0.0438	music
music_Tjurney_5	V, ethno	0.9427	music	0.0189	music
music_Tjurney_6	V, ethno	0.9135	speech	0.0194	music
music_Tgreen_1	V, orchestra	NaN	music	0.0189	music
music_Tgreen_2	V, orchestra	0.8957	speech	0.0200	music
music_Tgreen_3	V, orchestra	0.8646	speech	0.0217	music
music_Tgreen_4	V, orchestra	0.7809	speech	0.0277	music
music_Tgreen_5	V, orchestra	0.8777	speech	0.0246	music
music_Tgreen_6	V, orchestra	0.7819	speech	0.0276	music
music_Tgreen_7	V, orchestra	0.8198	speech	0.0350	music
music_Tend_1	V, orchestra	0.9672	music	0.0091	music
music_Tend_2	V, orchestra	0.8787	speech	0.0231	music
music_Tend_3	V, orchestra	NaN	speech	0.0188	music
music_Tend_4	V, orchestra	0.8536	speech	0.0189	music
music_Tend_5	V, orchestra	0.8808	speech	0.0271	music
music_Tend_6	V, orchestra	0.8794	speech	0.0286	music
music_Tend_7	V, orchestra	NaN	music	0.0252	music
music_Tbeach_1	V, orchestra	0.9372	music	0.0178	music
music_Tbeach_2	V, orchestra	0.9498	music	0.0182	music
music_Tbeach_3	V, orchestra	0.9370	music	0.0257	music
music_Tbeach_4	V, orchestra	0.9533	music	0.0191	music
music_Tbeach_5	V, orchestra	0.9515	music	0.0218	music
music_Tbeach_6	V, rock	0.9562	music	0.0222	music
music_Tbeach_7	V, rock	0.9435	music	0.0230	music
music_Tbeach_8	V, rock	0.9434	music	0.0227	music
music_Tbeach_9	V, spheric	0.6976	speech	0.0264	music
music_Tbeach_10	V, fx	0.9690	music	0.0087	music
music_Tbeach_11	V, fx	0.9500	music	0.0176	music
music_Tbeach_12	V, fx	0.8337	speech	0.0333	music
music_Tbeach_13	V, spheric	0.8009	speech	0.0329	music
music_Tbeach_14	V, spheric	0.9631	music	0.0163	music
music_smoothy_1	V, rock	0.8297	speech	0.0592	music
music_smoothy_2	V, hit	0.6740	speech	0.0813	music
music_smoothy_3	V, beat	0.8158	speech	0.0837	music
music_smoothy_4	V, voc	0.4116	speech	0.0896	music
music_sleepy_10s	V, spheric	NaN	music	0.0179	music
music_sleepy_5s	V, orchestra	0.9333	music	0.0223	music
music_sleepy_4	V, fx	0.6252	speech	0.0628	music
music_sleepy_3	V, orchestra	0.9034	speech	0.0231	music

music_sleepy_2	V, orchestra	0.9270	speech	0.0225	music
music_selina_1	V, ethno	0.9443	music	0.0299	music
music_selina_2	V, ethno	0.9569	music	0.0154	music
music_selina_3	V, ethno	0.9512	music	0.0392	music
music_selina_4	V, ethno	0.9122	speech	0.0618	music
music_selina_5	V, ethno	0.9076	speech	0.0375	music
music_selina_6	V, fx	0.8194	speech	0.0528	music
music_selina_7	V, ethno	0.9327	music	0.0341	music
music_selina_8	V, ethno	0.9484	music	0.0393	music
music_roy_1	V, schlager	0.9283	speech	0.0457	music
music_roy_2	V, schlager	NaN	music	0.0492	music
music_roy_3	V, schlager	0.9328	music	0.0434	music
music_roy_4	V, schlager	NaN	music	0.0417	music
music_roy_5	V, schlager	0.9237	music	0.0481	music
music_roy_6	V, schlager	NaN	music	0.0457	music
music_roy_7	V, schlager	0.9141	music	0.0378	music
music_roy_8	V, schlager	0.9474	music	0.0311	music
music_roy_9	V, schlager	0.9343	music	0.0341	music
music_roy_10	V, schlager	0.9307	music	0.0245	music
music_roy_11	V, schlager	0.9325	music	0.0213	music
music_roy_12	V, schlager	NaN	music	0.0359	music
music_roy_13	V, schlager	NaN	music	0.0323	music
music_pluto_1	V, fx	0.9617	music	0.0339	music
music_pluto_2	V, spheric	0.9457	music	0.0642	music
music_pluto_3	V, fx	0.9351	music	0.0645	music
music_pluto_4	V, beat	0.9283	speech	0.0519	music
music_pluto_5	V, beat	0.9238	speech	0.0412	music
music_pluto_6	V, beat	0.9394	music	0.0399	music
music_pluto_7	V, beat	0.9268	speech	0.0446	music
music_pluto_8	V, beat	0.9243	speech	0.0359	music
music_pluto_9	V, beat	0.8139	speech	0.2036	music
music_oos_1	V, synth	0.8894	speech	0.0932	music
music_oos_2	V, beat	0.7304	speech	0.1747	speech
music_oos_3	V, beat	0.7615	speech	0.1494	speech
music_oos_4	V, disco	0.8134	speech	0.0882	music
music_oos_5	V, disco	0.7275	speech	0.0941	music
music_oos_6	V, orchestra	0.8431	speech	0.0882	music
music_matrix_1	V, fx	0.9366	music	0.0569	music
music_matrix_2	V, beat	NaN	music	0.0429	music
music_matrix_3	V, fx	0.9187	speech	0.0521	music

music_matrix_4	V, beat	0.7108	speech	0.0824	music
music_matrix_5	V, fx	NaN	music	0.0361	music
music_matrix_6	V, fx	NaN	music	0.0590	music
music_matrix_7	V, fx	0.9232	speech	0.0576	music
music_matrix_8	V, beat	0.9444	music	0.0475	music
music_matrix_9	V, cut	0.0786	speech	0.0786	music
music_matrix_10	V, beat	0.9418	music	0.0481	music
music_matrix_11	V, fx	0.9538	music	0.0338	music
music_matrix_12	V, fx	0.9500	music	0.0363	music
music_matrix_13	V, fx	NaN	music	0.0429	music
music_matrix_14	V, orchestra	NaN	music	0.0495	music
music_matrix_15	V, fx	0.7993	speech	0.0545	music
music_kj_1	V, orchestra	0.8802	speech	0.1090	speech
music_kj_2	V, fx	0.8206	speech	0.0396	music
music_kj_3	V, orchestra	0.6522	speech	0.0511	music
music_kj_4	V, beat	0.8205	speech	0.0800	music
music_kj_5	V, song	0.7730	speech	0.0491	music
music_kj_6	V, beat	0.5837	speech	0.1566	speech
music_kj_7	V, beat	0.5024	speech	0.0413	music
music_kj_8	V, fx	0.8246	speech	0.0540	music
music_kj_9	V, voc	0.7034	speech	0.1719	speech
music_hal_1	V, spheric	0.9486	music	0.0419	music
music_hal_2	V, fx	0.9070	speech	0.0444	music
music_hal_3	V, beat	0.9180	speech	0.0433	music
music_hal_4	V, beat	0.9085	speech	0.0462	music
music_hal_5	V, beat	0.9082	speech	0.0464	music
music_hal_7	V, beat	0.9187	speech	0.0372	music
music_FA_7	CD, electro	0.8037	speech	0.0798	music
music_cnn_mn_1	V, orchestra	0.9496	music	0.0166	music
music_cnn_mn_2	V, beat	0.7090	speech	0.0760	music
music_bbc_sm_1	V, trumpet	0.7023	speech	0.0805	music
music_bbc_sm_2	V, flute	0.6050	speech	0.0477	music
music_bbc_sm_3	V, beat	0.8836	speech	0.0463	music
music_bbc_sm_4	V, beat	0.8886	speech	0.0325	music
music_bbc_bp_1	V, fx	0.8846	speech	0.0424	music
music_bbc_bp_2	V, beat	0.8027	speech	0.0410	music
speech_ddas_student	CD, s	0.7123	speech	0.1188	speech
speech_ddas_besuch	CD, s	0.7067	speech	0.1055	speech
speech_ddas_bewunderung	CD, s	0.7824	speech	0.0969	speech
speech_ddas_dinge	CD, s	0.6941	speech	0.0995	speech

speech_ddas_erleichterung	CD, s	0.7600	speech	0.0823	music
speech_ddas_flitterwochen	CD, s	0.7297	speech	0.1044	speech
speech_ddas_haus	CD, s	0.7580	speech	0.0782	music
speech_ddas_gewitter	CD, s	0.7493	speech	0.1127	speech
speech_ddas_haustor	CD, s	0.7461	speech	0.1044	speech
speech_ddas_hochzeit	CD, s	0.7493	speech	0.1127	speech
speech_ddas_kennen	CD, s	NaN	music	0.0969	speech
speech_ddas_lichter	CD, s	0.7406	speech	0.0836	music
speech_ddas_neutral	CD, s	0.7471	speech	0.0842	music
speech_ddas_unbesonnen	CD, s	0.6984	speech	0.1203	speech
speech_ddas_zauberin	CD, s	0.7524	speech	0.1162	speech
speech_dmdez_bekannte	CD, s	0.8102	speech	0.1096	speech
speech_dmdez_karten	CD, s	0.7239	speech	0.1282	speech
speech_dmdez_stadt	CD, s	0.7807	speech	0.1712	speech
speech_r2_friend	Video-, s, bs	0.9172	speech	0.0405	music
speech_r2_heart	Video-, s, bs	0.9055	speech	0.0400	music
speech_2days	Video-, s, bs	0.8913	speech	0.0334	music
speech_area	Video-, s, bs	0.8621	speech	0.0594	music
speech_beam	Video-, s, bs	0.9452	music	0.0301	music
speech_britain	Video-, s, bs	0.8423	speech	0.0827	music
speech_detectore	Video-, s, bs	0.9524	music	0.0278	music
speech_director	Video-, s, bs	0.7344	speech	0.0626	music
speech_earthquake	Video-, s, bs	0.8828	speech	0.0444	music
speech_end	Video-, s, bs	0.8939	speech	0.0548	music
speech_food	Video-, s, bs	0.8207	speech	0.0609	music
speech_frontline	Video-, s, bs	0.8455	speech	0.0598	music
speech_incident	Video-, s, bs	0.9088	speech	0.0257	music
speech_india	Video-, s, bs	0.7961	speech	0.0728	music
speech_people	Video-, s, bs	0.8179	speech	0.0355	music
speech_quake	Video-, s, bs	0.8379	speech	0.0584	music
speech_rescue	Video-, s, bs	0.9088	speech	0.0301	music
speech_security	Video-, s, bs	NaN	music	0.0358	music
speech_themselves	Video-, s, bs	NaN	music	0.0310	music
speech_forest_m	CD, s	0.6869	speech	0.0973	speech
speech_worm1_m	Video-, s, bs	0.8337	speech	0.0757	music
speech_worm2_m	Video-, s, bs	0.7908	speech	0.0609	music
speech_space_f	Video-, s, bs	0.8284	speech	0.0893	music
speech_french	Video-, s, bs	0.8165	speech	0.0648	music
speech_radio_fm	CD, s	0.8829	speech	0.0661	music
speech_holzhuetten	Video-, s, bs	0.6695	speech	0.0848	music

speech_hollywood_m	Video-, s, bs	0.6420	speech	0.0685	music
speech_foundation	Video-, s	0.7341	speech	0.0629	music
speech_Tmillion_1	Video-, s, bm	0.8110	speech	0.0549	music
speech_Tmillion_2	Video-, s, bm	0.9070	speech	0.0410	music
speech_Tmillion_3	Video-, s, bm	0.9476	music	0.0162	music
speech_Tmillion_4	Video-, s, bm	0.9190	speech	0.0246	music
speech_Tmillion_5	Video-, s, bm	0.9092	speech	0.0352	music
speech_Tmillion_6	Video-, s, bm	0.8651	speech	0.0295	music
speech_Tspiel_1	Video-, s, bm	0.7320	speech	0.0585	music
speech_Tspiel_2	Video-, s, bm	0.7542	speech	0.0582	music
speech_Tspiel_3	Video-, s, bm	0.8312	speech	0.0416	music
speech_Tspiel_4	Video-, s, bm	0.9252	speech	0.0224	music
speech_Tspiel_5	Video-, s, bm	0.8031	speech	0.0421	music
speech_Tjurney_1	Video-, s, bm	0.9296	speech	0.0178	music
speech_Tjurney_2	Video-, s, bm	0.8837	speech	0.0490	music
speech_Tjurney_3	Video-, s, bm	0.8799	speech	0.0354	music
speech_Tjurney_4	Video-, s, bm	0.6712	speech	0.0735	music
speech_Tjurney_5	Video-, s, bm	0.9681	music	0.0099	music
speech_Tjurney_6	Video-, s, bm	0.9253	speech	0.0291	music
speech_Tjurney_7	Video-, s, bm	0.9197	speech	0.0369	music
speech_Tgreen_1	Video-, s, bm	0.8181	speech	0.0392	music
speech_Tgreen_2	Video-, s, bm	0.8352	speech	0.0435	music
speech_Tgreen_3	Video-, s, bm	0.8682	speech	0.0151	music
speech_Tgreen_4	Video-, s, bm	0.9224	speech	0.0194	music
speech_Tend_1	Video-, s, bm	0.9448	speech	0.0161	music
speech_Tend_2	Video-, s, bm	0.9266	speech	0.0252	music
speech_Tend_3	Video-, s, bm	NaN	music	0.0340	music
speech_Tend_4	Video-, s, bm	0.8268	speech	0.0381	music
speech_Tend_5	Video-, s, bm	0.9171	speech	0.0251	music
speech_Tend_6	Video-, s, bm	0.9448	music	0.0221	music
speech_Tend_7	Video-, s, bm	0.9057	speech	0.0245	music
speech_Tend_8	Video-, s, bm	0.8745	speech	0.0261	music
speech_Tend_9	Video-, s, bm	0.9269	speech	0.0289	music
speech_Tend_10	Video-, s, bm	0.8943	speech	0.0257	music
speech_Tend_11	Video-, s, bm	0.8660	speech	0.0424	music
speech_Tend_12	Video-, s, bm	0.8905	speech	0.0325	music
speech_Tend_13	Video-, s, bm	0.8803	speech	0.0281	music
speech_Tend_14	Video-, s, bm	0.8115	speech	0.0320	music
speech_Tend_15	Video-, s, bm	0.9617	speech	0.0109	music
speech_Tend_16	Video-, s, bm	0.8317	speech	0.0362	music

speech_Tend_17	Video-, s, bm	0.8783	speech	0.0368	music
speech_Tend_18	Video-, s, bm	0.8605	speech	0.0256	music
speech_Tend_19	Video-, s, bm	0.8349	speech	0.0441	music
speech_Tend_20	Video-, s, bm	0.8456	speech	0.0411	music
speech_Tbeach_1	Video-, s, bm	0.9256	speech	0.0173	music
speech_Tbeach_2	Video-, s, bm	0.8608	speech	0.0428	music
speech_Tbeach_3	Video-, s, bm	0.9051	speech	0.0333	music
speech_Tbeach_4	Video-, s, bm	0.9232	speech	0.0268	music
speech_Tbeach_5	Video-, s, bm	NaN	speech	0.0437	music
speech_Tbeach_6	Video-, s, bm	0.8415	speech	0.0370	music
speech_Tbeach_7	Video-, s, bm	0.8562	speech	0.0355	music
speech_Tbeach_8	Video-, s, bm	0.8962	speech	0.0277	music
speech_Tbeach_9	Video-, s, bm	0.7916	speech	0.0530	music
speech_Tbeach_10	Video-, s, bm	0.8993	speech	0.0400	music
speech_Tbeach_11	Video-, s, bm	0.9596	music	0.0127	music
speech_smoothy_1	Video-, s, bm	0.8921	speech	0.0559	music
speech_smoothy_2	Video-, s, bs	0.9298	speech	0.0550	msuic
speech_sleepy_5s	Video-, s, bs	0.8580	speech	0.0410	music
speech_sleepy_2	Video-, s, bs	NaN	music	0.0491	music
speech_sleepy_3	Video-, s, bm	0.8337	speech	0.0524	music
speech_sleepy_4	Video-, s, bm	0.8492	speech	0.0490	music
speech_sleepy_5	Video-, s, bm	0.8698	speech	0.0405	music
speech_sleepy_6	Video-, s, bm	0.8594	speech	0.0428	music
speech_sleepy_7	Video-, s, bs	0.7819	speech	0.0526	music
speech_selina_1	Video-, s, bm	0.9356	music	0.0497	music
speech_selina_2	Video-, s, bm	0.8612	speech	0.0649	music
speech_selina_3	Video-, s, bm	0.9338	speech	0.0475	music
speech_selina_4	Video-, s, bm	0.9183	speech	0.0585	music
speech_selina_5	Video-, s, bm	0.9169	speech	0.0451	music
speech_selina_6	Video-, s, bm	0.7860	speech	0.0781	music
speech_roy_1	Video-, s	0.6795	speech	0.1642	speech
speech_roy_2	Video-, s	0.6587	speech	0.1421	speech
speech_roy_3	Video-, s	0.6786	speech	0.1562	speech
speech_roy_4	Video-, s	0.7145	speech	0.1469	speech
speech_roy_5	Video-, s	0.8585	speech	0.1008	speech
speech_pluto_1	Video-, s, bm	0.9360	music	0.0511	music
speech_pluto_2	Video-, s, bm	0.8248	speech	0.1570	music
speech_pluto_3	Video-, s, bm	0.9109	speech	0.0599	music
speech_pluto_4	Video-, s, bm	0.9239	speech	0.0550	music
speech_pluto_5	Video-, s, bm	0.9465	music	0.0219	music

speech_oos_1	Video-, s	0.7222	speech	0.9465	speech
speech_oos_2	Video-, s	0.5752	speech	0.2121	speech
speech_oos_3	Video-, s	0.7335	speech	0.1729	speech
speech_ntv_holz	Video-, s	0.6695	speech	0.0848	music
speech_matrix_1	Video-, s	0.5602	speech	0.2120	speech
speech_matrix_2	Video-, s	0.7428	speech	0.1449	speech
speech_matrix_3	Video-, s, bs	0.9220	speech	0.0621	music
speech_matrix_4	Video-, s	0.8753	speech	0.1028	speech
speech_matrix_5	Video-, s	0.8083	speech	0.1509	speech
speech_matrix_6	Video-, s, bm	0.8999	speech	0.0743	music
speech_matrix_7	Video-, s	0.7828	speech	0.1742	speech
speech_matrix_8	Video-, s, bm	0.8893	speech	0.0799	music
speech_matrix_9	Video-, s, bm	0.9258	speech	0.0583	music
speech_matrix_10	Video-, s, bm	0.8967	speech	0.0612	music
speech_kj_1	Video-, s	0.7730	speech	0.1163	speech
speech_kj_2	Video-, s	0.7544	speech	0.0990	speech
speech_kj_3	Video-, s	0.6588	speech	0.2170	speech
speech_kj_4	Video-, s	0.7258	speech	0.0993	speech
speech_kj_5	Video-, s, bm	0.7022	speech	0.1084	speech
speech_kj_6	Video-, s	0.4997	speech	0.1393	speech
speech_kj_7	Video-, s	0.7046	speech	0.1210	speech
speech_kj_8	Video-, s	0.6724	speech	0.1221	speech
speech_kj_9	Video-, s	0.1721	speech	0.1811	speech
speech_hal_1	Video-, s	0.8504	speech	0.1205	speech
speech_hal_2	Video-, s	0.9081	speech	0.0984	speech
speech_hal_3	Video-, s, bm	0.9382	music	0.0367	music
speech_hal_4	Video-, s	0.8345	speech	0.1285	speech
speech_hal_5	Video-, s	0.7927	speech	0.1506	speech
speech_cnn_n3c_1	Video-, s	0.8531	speech	0.0605	music
speech_cnn_mn_1	Video-, s, bm	0.8591	speech	0.0359	music
speech_cnn_mn_2	Video-, s, bm	0.9200	speech	0.0284	music
speech_cnn_mn_3	Video-, s	0.7765	speech	0.0617	music
speech_cnn_mn_4	Video-, s	0.7701	speech	0.0580	music
speech_cnn_mn_5	Video-, s	0.8571	speech	0.0624	music
speech_bbc_mn	Video-, s	0.6950	speech	0.1021	speech
speech_bbc_foun_1	Video-, s	0.6331	speech	0.0691	music
speech_bbc_foun_2	Video-, s	0.7330	speech	0.0744	music
speech_bbc_foun_3	Video-, s	0.8635	speech	0.0429	music
speech_bbc_bp_1	Video-, s, bm	0.6675	speech	0.1054	speech
speech_bbc_bp_2	Video-, s	0.6805	speech	0.0758	music

speech_bbc_bp_3	Video-, s	0.8108	speech	0.0898	music

Table D.1: Test results for audio content classification

Soundfile name ... name of the analyzed sound file

Source, style …… source of the file, music style / rhythm /characteristics

ser-value ……. value of the subband energy ratio

ser-class …….. classification result of the subband energy estimator

zcr-value ……. value of the standard deviation of the zero-crossings

zcr-class …….. classification result of the zero-crossing rate estimator

CD … soundfile taken from audio-CD

V ….. soundfile extracted from videofile, recorded via cinergy TV card

s ….. speech

bm ... background music

bn … background noise

bs … background sounds

fx … sound effects

Appendix E

Test results of uncoded audio content classification for audio sequences (music videos, music documentary, cinema trailers)

Table E.1 presents the test results of uncoded audio content classification for the case of audio scenes of an audio sequence, defined by audio scene detecting cut time points using the video tool described in chapter 6:

filename	filetype	cut frame number	content	zcr-classification	total
sylver	music video-	1	music	music	
		51	music	music	
		105	music	music	
		143	music	music	
		173	music	music	
		309	music	music	
		321	music	music	
		339	music	music	
		345	music	music	
		453	music	music	
		464	music	music	
		476	music	music	
		487	music	music	
		544	music	music	
		592	music	music	
		681	music	music	
		747	music	music	
		759	music	music	
		777	music	music	
		790	music	music	
		833	music	music	
		849	music	music	
		888	music	music	
		899	music	music	
		977	music	music	
		983	music	music	
		1.018	music	music	
		1.042	music	music	
		1.052	music	music	
		1.068	music	music	
		1.078	music	music	
		1.088	music	music	
		1.118	music	music	
		1.167	music	music	
		1.197	music	music	
		1.215	music	music	
		1.238	music	music	

1.256	music	music
1.268	music	music
1.282	music	music
1.288	music	music
1.323	music	music
1.336	music	music
1.365	music	music
1.395	music	music
1.419	music	music
1.436	music	music
1.454	music	music
1.472	music	music
1.492	music	music
1.504	music	music
1.515	music	music
1.542	music	music
1.562	music	music
1.570	music	music
1.585	music	music
1.647	music	music
1.666	music	music
1.684	music	music
1.700	music	music
1.720	music	music
1.726	music	music
1.736	music	music
1.750	music	music
1.756	music	music
1.812	music	music
1.843	music	music
1.864	music	music
1.876	music	music
1.892	music	music
1.906	music	music
1.922	music	music
1.928	music	music
1.934	music	music
1.959	music	music
1.993	music	music
2.007	music	music

2.030	music	music
2.074	music	music
2.098	music	music
2.127	music	music
2.162	music	music
2.187	music	music
2.233	music	music
2.245	music	music
2.257	music	music
2.269	music	music
2.293	music	music
2.313	music	music
2.335	music	music
2.347	music	music
2.373	music	music
2.413	music	music
2.467	music	music
2.491	music	music
2.511	music	music
2.533	music	music
2.556	music	music
2.570	music	music
2.598	music	music
2.678	music	music
2.706	music	music
2.725	music	music
2.737	music	music
2.755	music	music
2.809	music	music
2.839	music	music
2.863	music	music
2.893	music	music
2.907	music	music
2.925	music	music
2.953	music	music
2.971	music	music
2.983	music	music
3.014	music	music
3.026	music	music
3.082	music	music

3.097	music	music
3.203	music	music
3.246	music	music
3.258	music	music
3.270	music	music
3.311	music	music
3.463	music	music
3.480	music	music
3.500	music	music
3.517	music	music
3.597	music	music
3.644	music	music
3.677	music	music
3.849	music	music
3.872	music	music
3.888	music	music
3.900	music	music
3.914	music	music
3.926	music	music
3.939	music	music
4.015	music	music
4.021	music	music
4.081	music	music
4.140	music	music
4.199	music	music
4.253	music	music
4.265	music	music
4.363	music	music
4.445	music	music
4.493	music	music
4.525	music	music
4.540	music	music
4.563	music	music
4.601	music	music
4.625	music	music
4.639	music	music
4.667	music	music
4.679	music	music
4.691	music	music
4.745	music	music

		4.751	music	music	
		4.805	music	music	
		4.842	music	music	
		4.967	music	music	
		5.031	music	music	
		5.065	music	music	
		5.099	music	music	
		5.141	music	music	
		5.197	music	music	
		5.262	music	music	
		5.280	music	music	
		5.316	music	music	
		5.401	music	music	
		5.514	music	music	
		5.593	music	music	
		5.621	music	music	100%
pet shop boys	music video-	1	music / fade in	speech	
		154	music	music	
		201	music	music	
		248	music	music	
		274	music	music	
		345	music	music	
		366	music	music	
		486	music	music	
		524	music	music	
		574	music	music	
		616	music	music	
		643	music	music	
		663	music	music	
		689	music	music	
		715	music	music	
		737	music	music	
		759	music	music	
		812	music	music	
		854	music	music	
		893	music	music	
		922	music	music	
		952	music	music	
		985	music	music	
		1021	music	music	

1059	music	music
1109	music	music
1159	music	music
1190	music	music
1251	music	music
1299	music	music
1364	music	music
1405	music	music
1441	music	music
1476	music	music
1491	music	music
1525	music	music
1557	music	music
1625	music	music
1682	music	music
1721	music	music
1761	music	music
1804	music	music
1818	music	music
1859	music	music
1898	music	music
1917	music	music
2003	music	music
2058	music	music
2118	music	music
2150	music	music
2172	music	music
2187	music	music
2201	music	music
2217	music	music
2274	music	music
2312	music	music
2353	music	music
2394	music	music
2416	music	music
2441	music	music
2486	music	music
2556	music	music
2581	music	music
2609	music	music

2645	music	music
2671	music	music
2700	music	music
2729	music	music
2769	music	music
2814	music	music
2847	music	music
2876	music	music
2905	music	music
2938	music	music
2972	music	music
3022	music	music
3058	music	music
3125	music	music
3188	music	music
3221	music	music
3258	music	music
3334	music	music
3361	music	music
3390	music	music
3433	music	music
3533	music	music
3576	music	music
3589	music	music
3611	music	music
3672	music	music
3723	music	music
3776	music	music
3839	music	music
3875	music	music
3941	music	music
3990	music	music
4042	music	music
4077	music	music
4103	music	music
4140	music	music
4193	music	music
4245	music	music
4305	music	music
4337	music	music

4363	music	music
4388	music	music
4422	music	music
4460	music	music
4500	music	music
4573	music	music
4633	music	music
4678	music	music
4737	music	music
4748	music	music
4766	music	music
4824	music	music
4862	music	music
4904	music	music
4938	music	music
4974	music	music
5002	music	music
5053	music	music
5105	music	music
5152	music	music
5196	music	music
5232	music	music
5292	music	music
5334	music	music
5457	music	music
5527	music	music
5555	music	music
5666	music	music
5693	music	music
5748	music	music
5785	music	music
5825	music	music
5851	music	music
5876	music	music
5935	music	music
5976	music	music
6014	music	music
6039	music	music
6082	music	music
6114	music	music

		6131	music	music	
		6155	music	music	
		6195	music	music	
		6225	music	music	
		6363	music	music	
		6401	music	music	
		6453	music	music	
		6511	music	music	
		6585	music	music	
		6629	music	music	
		6671	music	music	
		6717	music	music	
		6746	music	music	
		6771	music	music	
		6829	music	music	
		6865	music	music	
		6902	music	music	
		6971	music	music	
		7028	music	music	
		7096	music	music	
		7153	music	music	
		7187	music	music	
		7230	music	music	
		7298	music	music	
		7385	music	music	
		7455	music	speech	100% without lead fx's
		7563	music/fade out	speech	98.8235 with lead fx's
roy black	music doc.	1	speech	speech	
		32	speech	speech	
		322	music	music	
		390	music	music	
		435	music	music	
		578	music	music	100%
angel_end	music doc.	1	music	music	
		46	music	music	
		306	speech	speech	
		460	speech	speech	100%
come_undone	music doc.	1	speech	speech	

		153	speech	speech	
		406	music	music	
		425	music	music	
		585	speech	speech	
		645	speech	speech	
		675	speech	music	
		704	speech	speech	
		764	speech	speech	89%
angel_start	music doc.	1	speech	speech	
		91	music	speech	
		150	music	music	
		210	music	music	
		280	music	music	
		300	music	music	
		324	speech	speech	
		349	music	music	
		369	music	music	
		387	music	music	90%
joe black	cinema trailer	1	silence	music	
		20	music / fade in	speech	
		80	music	music	
		94	music	music	
		127	music	music	
		160	music	music	
		191	music	music	
		209	music	music	
		245	music	music	
		268	music	music	
		354	music	music	
		408	music	music	
		453	music	music	
		485	music	music	
		514	music	music	
		561	music	music	
		594	music	music	
		649	music	music	
		676	music	music	
		695	music	music	
		709	music	music	
		742	music	music	

756	music	music
795	music	music
821	music	music
869	music	music
937	music	music
981	music	music
1082	music	speech
1137	music	speech
1217	music	speech
1316	music	music
1366	music	music
1418	music	music
1451	music	music
1485	music	music
1534	music	music
1578	music	music
1647	music	music
1699	music	music
1775	music	music
1807	music	music
1841	music	music
1915	music	music
1982	music	music
2013	music	speech
2048	music	music
2113	music	music
2220	music	music
2253	music	music
2331	music	music
2377	music	music
2420	music	music
2447	music	music
2477	music	music
2496	music	music
2527	music	music
2555	music	music
2584	music	music
2616	music	music
2644	music	music
2675	music	speech

2712	music	speech
2741	music	music
2776	music	music
2814	music	music
2832	music	music
2874	speech	speech
2903	speech	speech
2932	music	speech
2962	music	music
2996	music	music
3024	music	music
3049	music	music
3119	music	music
3146	music	music
3180	music	music
3237	music	music
3264	music	music
3304	music	music
3319	music	music
3354	music	music
3378	music	music
3430	music	music
3464	music	music
3509	music	music
3545	music	music
3593	music	music
3620	music	music
3659	music	music
3689	music	music
3725	music	music
3747	music	music
3791	music	music
3814	music	music
3852	music	music
3887	music	music
3933	music	music
3965	music	music
4004	music	music
4038	music	music
4076	music	music

		4088	music	music	
		4104	music	music	
		4132	music	music	
		4152	music	music	
		4202	music	music	
		4249	music	music	
		4419	music	music	
		4567	music	music	
		4604	music	music	
		4687	music	music	
		4718	music	music	
		4875	music	music	
		4928	music	music	
		4991	music	music	
		5086	music	music	
		5250	music	music	
		5284	music	music	94.8276 without lead fx's
		5335	music/fade out	speech	92.437 with lead fx's
das spiel	cinema trailer	1	music	music	
		5	music	music	
		37	music	music	
		47	music	music	
		94	music	music	
		143	music	music	
		193	music	music	
		245	music	music	
		263	music	music	
		307	music	music	
		339	music	music	
		402	music	music	
		432	music	music	
		472	music	music	
		488	music	music	
		520	music	music	
		555	music	music	
		580	music	music	
		599	music	music	
		610	music	music	

630	music	music
685	music	music
718	music	music
736	music	music
763	music	music
792	music	music
868	music	music
910	music	music
940	music	music
961	music	music
983	music	music
993	music	music
1035	music	music
1083	music	music
1117	music	music
1165	music	music
1199	music	music
1214	music	music
1254	music	music
1271	music	music
1294	music	music
1342	music	music
1372	music	music
1407	music	music
1439	music	music
1484	music	music
1511	music	music
1522	music	music
1535	music	music
1551	music	music
1588	music	music
1627	music	music
1679	music	music
1701	music	music
1727	music	music
1739	music	music
1768	music	music
1803	music	music
1823	music	music
1869	music	music

		1884	music	music	
		1895	music	music	
		1909	music	music	
		1933	music	music	
		1962	music	music	
		2019	music	music	
		2069	music	music	
		2094	music	music	
		2119	music	music	
		2160	music	music	
		2190	music	music	100%
smoothy	cinema trailer	1	silence	music	
		9	music	music	
		93	music	music	
		121	music	music	
		148	music	music	
		161	speech	music	
		200	speech	speech	
		224	speech	music	
		270	speech	music	
		341	music	music	
		411	music	music	
		444	music	music	
		465	music	speech	
		481	music	music	
		503	music	music	
		517	music	music	
		553	music	music	
		563	music	music	
		574	music	music	
		678	music	music	
		791	music	music	
		823	music	music	
		883	music	music	
		905	music	music	
		960	music	music	
		1017	music	music	
		1052	music	music	
		1071	music	music	
		1087	music	music	

1125	music	music
1209	music	music
1229	music	music
1248	music	music
1280	music	music
1287	music	music
1307	music	music
1350	music	music
1385	music	music
1424	music	music
1448	music	music
1463	music	music
1488	music	music
1515	music	music
1569	music	music
1606	music	music
1650	music	music
1662	music	music
1691	speech	speech
1752	music	music
1783	speech	music
1829	speech	speech
1854	music	music
1878	music	music
1902	music	music
1973	music	music
2025	music	music
2067	music	music
2085	music	music
2108	music	music
2139	music	music
2170	music	music
2192	music	music
2263	music	music
2320	music	music
2350	music	music
2373	music	music
2417	music	music
2441	music	music
2469	music	music

	2502	music	music	
	2528	music	music	
	2541	music	music	
	2572	music	music	
	2612	music	music	
	2625	music	music	
	2645	music	music	
	2694	music	music	
	2729	music	music	
	2754	music	music	
	2783	music	music	
	2812	music	music	
	2841	music	music	93.75% without silence
	2934	silence	speech	
out of sight	1	music / fade in	speech	
	83	music	music	
	198	music	speech	
	311	music	speech	
	375	music	speech	
	386	music	speech	
	397	music	speech	
	433	music	music	
	528	music	music	
	540	music	music	
	673	music	music	
	727	music	music	
	748	music	music	
	767	music	music	
	788	music	music	
	807	music	music	
	829	music	music	
	847	music	speech	
	892	speech	speech	
	934	speech	speech	
	971	speech	speech	
	997	speech	speech	
	1016	speech	speech	
	1039	speech	speech	
	1082	speech	speech	

1102	music	music
1142	music	music
1161	music	music
1177	music	music
1201	music	music
1213	music	music
1223	music	music
1243	music	music
1254	music	music
1264	music	music
1291	music	music
1304	music	music
1333	music	music
1343	speech	speech
1353	speech	speech
1371	music	music
1381	music	music
1389	music	music
1398	music	music
1420	music	music
1432	speech	speech
1447	music	music
1599	music	music
1686	music	music
1764	music	music
1808	music	music
1821	music	music
1833	music	music
1859	music	music
1881	music	music
1904	music	music
1919	speech	speech
1933	speech	speech
1960	speech	speech
1974	music	music
1987	music	music
2034	music	music
2061	music	music
2117	music	music
2139	music	music

	2168	music	music	
	2188	music	music	
	2198	music	music	
	2213	music	music	
	2228	music	music	
	2239	music	music	
	2267	music	music	
	2304	music	music	
	2318	music	music	
	2356	music	music	
	2376	music	music	
	2400	music	music	
	2419	music	music	
	2439	music	music	
	2471	music	music	
	2493	music	music	
	2505	music	music	
	2519	music	music	
	2541	music	music	
	2558	music	music	
	2577	music	speech	
	2616	music	music	
	2637	music	music	
	2725	music	music	
	2869	music	music	
	2979	music	music	92.135% without lead fx's
	3210	music/fade out	speech	
pluto nash	1	music/fade in	speech	
	105	music	music	
	350	music	music	
	397	music	music	
	580	music	music	
	631	music	music	
	679	music	music	
	696	music	music	
	742	music	music	
	764	music	music	
	782	music	music	
	800	music	music	

818	music	music
844	music	music
877	music	music
895	music	music
919	music	music
936	music	music
960	music	music
990	music	music
1056	music	music
1119	speech	speech
1133	music	music
1155	music	music
1170	music	music
1196	music	music
1207	music	music
1222	music	music
1233	music	music
1246	music	music
1271	music	music
1348	music	music
1378	music	music
1398	music	music
1421	music	music
1481	music	music
1518	music	music
1559	music	music
1572	music	music
1596	music	music
1612	music	music
1635	music	music
1659	music	music
1730	music	music
1793	music	music
1852	music	music
1885	music	music
1907	music	music
1928	music	music
1944	music	music
1979	music	music
1990	music	speech

		2048	speech	speech	
		2109	speech	speech	
		2181	speech	speech	
		2239	speech	speech	
		2274	speech	speech	
		2288	music	music	
		2307	music	music	98.2456% without lead fx's
matrix_usa	cinema trailer	1	music	music	
		58	music	music	
		132	music	music	
		139	music	music	
		218	music	music	
		233	music	music	
		252	music	music	
		282	music	music	
		293	music	music	
		308	music	music	
		344	music	music	
		425	music	music	
		464	music	music	
		525	music	music	
		538	music	music	
		562	music	music	
		577	music	music	
		607	music	music	
		633	music	music	
		662	music	music	
		679	music	music	
		693	music	music	
		703	music	music	
		713	music	music	
		732	music	music	
		764	music	music	
		779	music	music	
		793	music	music	
		819	music	music	
		845	music	music	
		858	music	music	
		870	music	music	

	885	music	music	
	913	music	music	
	962	music	music	
	977	music	music	
	993	music	music	
	1059	music	music	
	1079	music	music	
	1095	music	music	
	1112	music	music	
	1137	music	music	
	1153	music	music	
	1173	music	music	
	1203	music	music	
	1217	music	music	
	1273	music	music	
	1304	music	music	
	1354	music	music	
	1379	music	music	
	1432	music	music	
	1457	music	music	
	1473	music	music	
	1487	music	music	
	1507	music	music	
	1538	music	music	
	1572	music	music	
	1592	music	music	
	1604	music	music	
	1663	music	music	
	1682	music	music	
	1697	music	music	
	1747	music	music	
	1762	music	music	100%
matrix_german cinema trailer	1	music / fade in	speech	
	177	music	music	
	201	music	music	
	212	music	music	
	224	music	music	
	246	music	music	
	257	music	music	
	270	music	music	

293	music	music
304	music	music
317	music	music
339	music	music
350	music	music
363	music	music
387	music	music
398	music	music
409	music	music
432	music	music
443	music	music
456	music	music
478	music	music
501	music	music
515	music	music
571	music	music
590	music	music
614	speech	speech
647	speech	speech
658	speech	speech
677	music	speech
725	music	music
758	music	music
766	music	music
848	music	music
866	music	music
920	music	speech
945	speech	speech
983	speech	speech
1008	speech	speech
1041	music	music
1070	music	speech
1108	music	music
1122	music	music
1146	music	music
1165	music	music
1196	music	music
1209	music	music
1248	music	music
1261	music	music

1277	music	music
1315	music	music
1328	music	music
1346	speech	speech
1362	music	speech
1380	music	speech
1401	music	music
1419	speech	speech
1437	music	music
1499	music	music
1520	music	music
1538	music	music
1563	speech	speech
1594	speech	speech
1618	speech	speech
1626	speech	speech
1667	speech	speech
1708	speech	speech
1729	speech	speech
1740	speech	speech
1751	music	music
1773	music	music
1794	music	music
1806	speech	music
1819	speech	speech
1832	music	music
1857	music	music
1889	speech	speech
1951	music	music
1971	music	music
2043	music	music
2062	music	music
2077	music	music
2087	music	music
2097	music	music
2117	music	music
2147	music	music
2167	music	music
2202	music	music
2212	music	music

2236	music	music
2263	music	music
2275	music	music
2294	music	music
2320	music	music
2390	music	music
2402	music	music
2440	music	music
2453	music	music
2466	speech	speech
2542	music	music
2617	music	music
2626	music	speech
2688	music	music
2699	music	music
2747	music	music
2770	music	music
2813	music	music
2834	music	music
2854	music	music
2873	music	music
2894	music	music
2935	music	music
2955	speech	speech
2975	music	music
3037	music	music
3055	music	music
3080	music	music
3134	music	music
3155	music	music
3194	music	music
3276	music	music
3295	music	music
3326	music	music
3356	music	music
3375	music	music
3395	music	music
3415	music	music
3435	music	music
3453	music	music

	3476	music	music	
	3496	music	music	
	3516	music	music	
	3537	music	music	
	3556	music	speech	
	3576	music	music	
	3596	music	music	
	3602	music	music	
	3616	music	music	
	3636	speech	speech	
	3697	speech	speech	
	3726	speech	speech	
	3736	speech	speech	
	3748	speech	speech	
	3763	music	music	
	3777	music	music	
	3846	music	music	
	3870	music	music	
	3979	music	music	
	4018	Music	music	
	4035	Music	music	
	4075	Music	music	95.2381% without lead fx's
	4137	music/fade out	speech	

Table E.1: Test results for music videos, music documentations, cinema trailers.

Appendix F

Abbreviations

3GPP	Third generation partnership project
A_ind, D_ind	Disturbance indicators
AAC	Advanced Audio Codec
AD	Auditory Distance
AMR	Adaptive Multi Rate
AMR NB	Adaptive Multi Rate Narrowband
AMR WB	Adaptive Multi Rate Wideband
AQ	Audio Quality
DRM	Digital Rights Management
EDGE	Enhanced Data Rate for GSM Evolution
FFT	Fast Fourier Transformation
GERAN	GSM / EDGE-based RAN
GSM	Global System for Mobile Communications
HTML	Hyper Text Marker Language
IFD	Integrated Frequency Distance
IP	Internet Protocol
ITU	International Telecommunications Union
MMS	Multimedia Session Service
MNB	Measurement Normalizing Block Technic
MOS	Mean Opinion Score
MOV's	Model Output Variables
ODG	Objective Difference Grade
OMOS	Objective Mean Opinion Score
PAMS	Perceptual Analysis Measurement System
PDP	Packet Data Protocol
PEAQ	Perceptual Evaluation of Audio Quality
PESQ	Perceptual Evaluation of Speech Quality

PSQM	Perceptual Speech Quality Measurement
PSQM+	Advanced Speech Quality Measurement
PSS	Packet Switched Streaming Service
QoS	Quality of Service
RFC	IETF Request For Comments
RTSP	Real Time Streaming Protocol
SDP	Session Description Protocol
UMTS	Universal Mobile Telecommunications System
URI	Universal Resource Identifier
UTRAN	UMTS Radio Access Network
VoIP	Voice over IP
VQEP	Video Quality Expert Group
WAP	Wireless Application Protocol
WWW	World Wide Web
ZCR	Zero Crossing Rate

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