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Subjective and Objective Comparative Study of DAB+ Broadcast System

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Broadcasting services seek to optimize their use of bandwidth in order to maximize user's quality of experience. They aim to transmit high-quality digital speech and music signals at the lowest bitrate. They intend to offer the best quality under available conditions. Due to bandwidth limitations, audio quality is in conflict with the number of transmitted radio programs. This paper analyzes whether the quality of real-time digital DAB+ broadcasted radio programs surpasses the quality offered by analog FM radio. We also perform a subjective and objective quality assessment comparative study of the HE-AAC v2 audio codec used in DAB+. The subjective studies were carried out using the MOS test methodology, whereas the objective tests were realized using the ViSQOLAudio metric. These studies were followed by a questionnaire concerning the migration from analog to digital radio domain.

Keywords: audio quality assessment; broadcasting; digital audio broadcasting; quality of experience; quality of service; radio communication.

1. Introduction

Broadcasting systems generally consist of different signal processing blocks. This signal processing, e.g. source coding and channel coding, may utilize different codecs and bitrates which, as a result, have a significant impact on the end user perceived quality (COST Action IC1003, 2012). Therefore, it is important to study how different coding schemes affect the QoS (*Quality of Service*) and QoE (*Quality of Experience*), especially under limited bandwidth resources.

In the age of digital media, when it comes to delivering audio content, the target is to distribute as many radio programs as possible within limited bandwidth resources. Of course, the higher the bitrate, the higher the quality and user satisfaction. An appropriate balance between the number of broadcasted radio programs and their bitrate is a delicate, yet vital decision. Bitrate assignment is still a widely discussed topic (Brachmański, Kin, 2013; Kin, 2013).

Certainly, digital distribution of audio content is also realized by cellular systems, such as 3G UMTS (*Universal Mobile Telecommunication System*) (ETSI TS 136 300, 2008), as well as WLAN (*Wireless Local Area Network*) (IEEE, 2012). Some telecoms offer unlimited transfer for a predefined number of streaming platforms and services. However, streaming multimedia puts much stress not only on the telecom infrastructure, but also on mobile devices such as smartphones or tablets, which is clearly visible in the decrease of battery life. It should be emphasized that terrestrial broadcasting provides the same quality for each person, regardless of the number of simultaneous users.

2. Quality of experience

It is assumed that high performance and transmission quality will lead to high acceptance and usability of offered services, clearly visible in the growth of users. However, low quality or best effort services such as SMS (*Short Message Service*) or e-mails have gained enormous popularity. This clearly shows that the relationship between performance, quality and acceptance of a service is not fully recognized. Also the term quality can be understood in many different ways.

Engineers perceive the term as quality of service, which is a synonym for network performance and reliability. But does this term, generally describing the characteristics of machines, devices and their parameters, really reflect the process of perception of a human individual? According to (Jekosch, 2005), quality can be defined from a person's point of view, which involves a process of comparing perceptual events with a known reference. The user's previous experience may in fact influence the opinion about what is actually perceived.

That is why the term quality of experience has gained interest. It focuses mainly on defining the characteristics of media transmission systems or services and their acceptance by customers. Broadcasters, content providers and network operators no longer intend to deliver services with simply high QoS, but with satisfying QoE to their customers.

When examining the interaction between human-machine interfaces, a shift can be observed. Issues such as effectiveness and efficiency, also referred to as usability, tend towards a term User Experience (UX), related with the experience people have when using various interfaces. Clearly, all issues related with QoS, QoE and UX need further study in order to really describe what does the term quality mean for the typical user.

Since 2011, the European Network on Quality of Experience in Multimedia Systems and Services (COST Action IC 1003) (QUALINET), started the scientific discussion about the definition of the term quality. This multidisciplinary group, focused on quality aspects of different multimedia services, approached this problem from a different perspective. They started to extend the notion of network-centric QoS and user-centric QoE. The main scientific objective is to develop new subjective methodologies and instrumental quality metrics that would keep up with new trends in multimedia communication systems.

3. Broadcast quality

From a broadcaster's perspective, the term quality is a key factor for evaluating systems and services. During both the design and operation phase, the main objective is to create certain types of new experiences, that would interest the potential user. Whenever a new service is introduced, whether a substitute or intended competition, it should offer features that are unique and distinguish them from other services available on the market.

Currently, digital audio broadcasting is a well-established method for consuming content. Systems like DAB (*Digital Audio Broadcasting*) (ETSI EN 300 401, 2006), DRM (*Digital Radio Mondiale*) (ETSI ES 201 980, 2014) or T-DMB (*Terrestrial Digital Multimedia Broadcasting*) (Cho *et al.*, 2007) are intended to replace traditional analog AM and FM radio. Thanks to the popularity and availability of mobile and portable devices, streaming and webcasting services allow users to listen to music almost anytime and everywhere. However, network bandwidth constraints are viable across the diverse range of techniques used. As a result, content providers and broadcasters must support a wide range of codecs and bitrates in order to optimize the user perceived QoE. Studies show that a bitrate of 256 kbps can deliver lossy compressed content that is indistinguishable from the uncompressed original file (Hines *et al.*, 2010). Psychoacoustic inspired compression schemes utilize signals that are optimized from the perspective of the human auditory system (Dobrucki, Kozłowski, 2010). Efficient bandwidth management can not only enhance user experience. Any bandwidth savings can be used to introduce another service.

In case of digital broadcasting systems such as DAB+ (*Digital Audio Broadcasting plus*) (ETSI TS 102 563, 2010), the mere perceived quality is a mixture of different parameters such as: delay, latency, channel impulse response, quantization noise or SNR (*Signal-to-Noise Ratio*). However, the most significant aspects are bandwidth limitations which have a significant impact on the assigned bitrate of a radio program. Additional information concerning quality aspect in broadcasting and webcasting systems can be found in (Gilski, Stefański, 2016).

DAB+ was designed as a substitute or replacement for the well-known analog FM radio. It has a lot to offer, besides transmitting audio signals. The main factors that attract new users to this service are:

- a clearly noticeable higher transmission quality,
- a stable reception, especially in case of mobile or motorized users,
- easy to operate receivers with a number of additional services.

After analyzing the available literature, three broadcast quality criteria can be easily distinguished (Kozamernik, 1997; Berg *et al.*, 2013):

1) clarity and transparency of the audio material – this parameter particularly refers to the early days of radio transmission,
2) broadcast quality – the overall quality is ranked as >4.0 in the MOS (*Mean Opinion Score*) or >80 in the MUSHRA (*MUltiple Stimulus with Hidden Reference and Anchor*) scale,
3) FM quality – the digital standard, intended to become a substitute or replacement of analog transmission, should provide comparable or higher quality.

In this paper, we focus on determining whether the current quality of DAB+ can compete with the quality offered by existing FM radio.

3.1. Subjective quality metrics

The most reliable method for quality assessment is via subjective testing with a group of listeners. The ITU provided a widely used recommendation (ITU P.800, 1996) defining the procedure of quality tests. The most frequently used is MOS (ITU BS.1116-1, 1997) where listeners rate the quality in a 5-step scale from 1 (bad quality) to 5 (excellent quality). Recently, a new methodology called MUSHRA (ITU BS.1534-1, 2003), has been gaining popularity. MUSHRA allows listeners to compare treatments and rank them on a continuous scale from 0 to 100. They are presented with a labeled reference signal and a number of unlabeled test samples, so-called stimuli, including 1 or 2 anchors, being a 3.5 kHz or 7 kHz LPF-processed version of the reference signal. Listeners are asked to rank each sample from 0 to 100 in 5-step intervals: bad (0–20), poor (21–40), fair (41–60), good (60–80), and excellent (81–100). Biases in MUSHRA have been investigated (Zieliński, 2015), but this methodology has been used in a variety of tests showing a good ability to rank low bitrate codecs.

Of course, MOS or MUSHRA scores can vary, based on cultural or language issues, number of listeners, or even test conditions. That is why usually the range of tested audio samples is limited, depending on the interest for a specific research topic. Compared with objective testing automated by software, subjective testing is viewed as expensive and time consuming. As a result, objective test metrics have been developed and remain a topic of active research.

3.2. Objective quality metrics

Broadcasters and telecomms want to evaluate the quality of speech and music signals of offered services. This is a crucial issue for the whole process of planning as well as implementation, monitoring and maintenance purposes.

Objective metrics can be classified into two main categories: parameter-based and signal-based methods. Parameter-based methods do not test signals over the channel but instead predict the speech quality through modelling the channel parameters. On the other hand, signal-based methods predict the quality based on evaluation of a test signal at the output of the channel. Signal-based methods can be further divided into two subcategories: intrusive and non-intrusive methods.

Intrusive methods use an original reference signal and compare it with a degraded signal, representing the output signal of the tested system. The PSQM (*Perceptual Speech Quality Measure*) (ITU P.861, 1998) was the first attempt to model a human listener and predict the perceived quality. Later on, new objective quality metrics for speech and music signals have emerged, including PESQ (*Perceptual Evaluation of Speech Quality*) (ITU P.862, 2001) and PEAQ (*Perceptual Evaluation of Audio Quality*) (ITU BS.1387, 2001), which allow to predict the quality by comparing a reference signal to a received signal.

PESQ was developed first to provide an objective estimate of narrowband speech, and was later extended for wideband speech (ITU P.862.2, 2007). On the other hand, PEAQ has two versions, one optimized for speech and the other that adds a filterbank-based ear model to the basic FFT-based model in order to improve accuracy. Both versions produce the output in the form of ODG (*Objective Difference Grade*) quality score, which is an objective approximation of the subjective difference grade used to determine small audio impairments.

A decade later, ITU standardized a new objective metric, POLQA (*Perceptual Objective Listening Quality Assessment*) (ITU P.863, 2014). It was initially designed for speech quality assessment for VoIP (*Voice over IP*) in narrowband mode (300–3400 Hz) or super-wideband (50–14 000 Hz) mode. However, it proved to be quite accurate as an audio quality model (Pinson *et al.*, 2013).

In comparison to intrusive methods, one of the most widely used models is ANIQUE+ (*Auditory Non-Intrusive Quality Estimation plus*) (ANSI, 2006), a signal-based method used to predict the quality without access to any reference signal. The ITU has also introduced its recommendation (ITU P.563, 2004). It should be noticed that this topic is still an active area of research.

Recently, an alternative speech quality model, called ViSQOL (*Visual Speech Quality Objective Listener*) (Hines *et al.*, 2015), has been developed for quality assessment in narrowband (150–3400 Hz) and wideband (50–8000 Hz) mode. It is a full-reference speech metric, comparable to POLQA, that uses similarity between spectrograms to measure quality. In this paper we analyzed ViSQOLAudio (Hines *et al.*, 2015), a fullband adaptation used for audio quality evaluation. The latest update enables to assess the quality of both speech and music signals in a 5-step MOS scale. An extensive review of objective speech quality models can be found in (Möller *et al.*, 2011).

4. Quality assessment study

For broadcasting systems, whether talking about development or maintenance purposes, quality needs to be assessed in a reliable way. Of course, subjective study with human listeners is the most valuable and reliable measurement for speech or music quality. However, it is time consuming and expensive to perform. On the other hand, objective measures aim to model this assessment using software and hardware to give estimates comparable with subjective judgements.

In available literature there are publications concerning both objective and subjective quality assessment or coding efficiency, i.e. (Meltzer, Moser, 2006; Počta, Beerends, 2015). These studies focus on analyzing a selected number of audio signal samples, that were processed using different source codecs and/or different bitrates. Despite diligent search, we did not encounter studies concerning real-time radio broadcasted material, particularly the quality of DAB+ and FM transmission. That is why we decided to carry out our experiment.

4.1. About the test

Currently, the DAB+ multiplex in Poland offers 9 radio programs transmitting speech and music signals, as well as 2 additional data services: Data and Journaline. For the purpose of this test, we have initially selected 5 radio programs that are simulcasted in both FM and DAB+, in order to check whether the transmission quality of digital radio can compete with the quality offered by analog radio. The profile and bitrate of each simulcasted radio station are shown in Fig. 1.

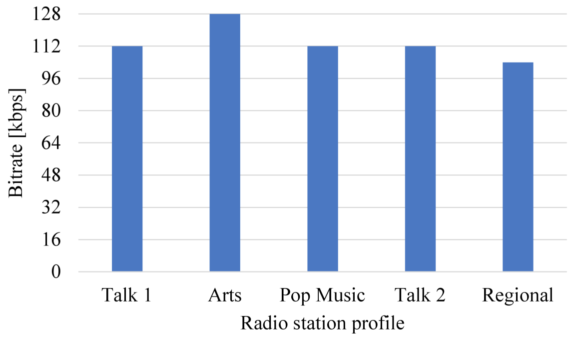

Fig. 1. Profile and bitrate of radio stations simulcasted in both FM and DAB+.

Currently, there is one radio program dedicated to classical music (Arts), two stations of a general profile (Talk 1 and Talk 2), one program transmitting particularly popular music (Pop Music) and one regional broadcasting station (Regional).

However, concerning the broad range of radio programs, their profiles and target groups, as well as the number of different music genres, we decided to expand the study with an analysis of signal samples processed using the HE-AAC v2 (*High Efficiency Advanced Audio Coding*) codec utilized in DAB+.

The test was performed on a group of 45 people, aged between 18 and 25 years old. None of them had hearing disorders. Listeners were not informed about the actual bitrate of a particular program which could affect the quality assessment study. They were only informed about the profile of the current ranked radio station. The study, involving both subjective and objective quality assessment, consisted of 4 parts:

1) subjective quality assessment of real-time DAB+ broadcasted radio programs,
2) subjective comparative quality assessment of DAB+ radio programs with respect to simulcasted FM programs,
3) objective and subjective quality assessment of signal samples processed with the AAC codec at different bitrates,
4) questionnaire concerning the switchover from analog to digital radio domain.

The subjective tests were carried out according to recommendation (ITU BS.1284, 2003) with a short break between each part, whereas the objective tests were performed using the ViSQOLAudio algorithm.

Each person assessed the quality individually and was informed about the aim and test scenario. All participants took a training phase before starting the essential listening test in order to learn the functionality of the user interface and become familiar with the listening equipment. Tests were performed in turns, one individual after another. A single session took approx. 20–25 minutes. Listeners were allowed to adjust the volume according to their preferences. Tests were conducted using AKG K550 closed-back headphones, as measuring with loudspeakers introduced a risk that room acoustics could influence the results.

4.2. Subjective quality assessment of real-time DAB+ broadcasted radio programs

Listeners were asked to rate the overall quality of each real-time transmitted DAB+ radio program, over a period of approx. 10–20 s, separated by a 1 s interval, in a five-step MOS ACR (*Absolute Category Rating*) scale from 1 (bad quality) to 5 (excellent quality). The subjective results are shown in Fig. 2.

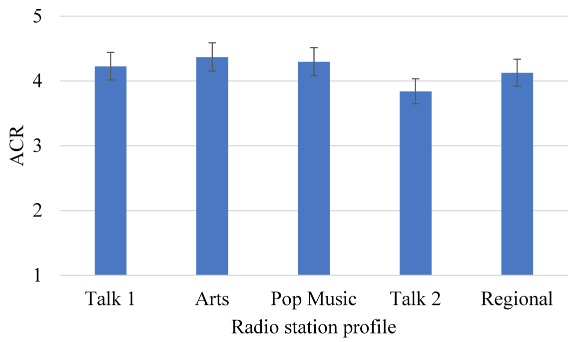

Fig. 2. Subjective quality assessment of real-time DAB+ broadcasted radio programs.

According to the results, the overall quality was ranked as good. This can be viewed as a confirmation that the bitrate, assigned for each individual program, was chosen appropriately. Since digital radio is less vulnerable to multipath effects and noise, the DAB+ broadcasted programs were reported as free from interference. This means that currently both the QoS and QoE aspects meet user expectations.

4.3. Subjective comparative quality assessment of DAB+ radio programs with respect to simulcasted FM programs

Next, each individual was asked to compare the quality of DAB+ and FM broadcasted programs. They were asked to rate the DAB+ audio material (sample "B") with respect to FM audio material (sample "A") in a 7-step MOS CCR (*Comparative Category Rating*) scale from −3 ("A" is much better than "B") to +3 ("B" is much better than "A").

The samples were presented to the listeners in a single "A–B" pair, over a period of approx. 10–20 s, separated by a 0.5–1 s interval. Each pair, representing the same simulcasted radio station, was rated separately. The subjective results of this comparative study are shown in Fig. 3.

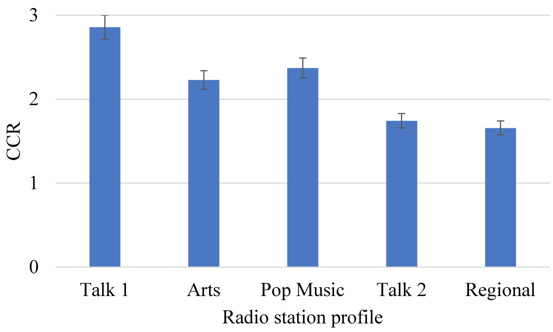

Fig. 3. Subjective comparative quality assessment of DAB+ radio programs with respect to simulcasted FM programs.

According to the obtained results, in case of each DAB+ radio program there was a clearly noticeable increase in quality. This indicates that the current quality offered by DAB+ surpasses that of FM.

Furthermore, the CCR method can be viewed as more accurate than ACR because it enables listeners to distinguish little differences between two audio samples. It can be noted that when comparing digital DAB+ with analog FM transmission, the relation between bitrate and quality evaluation is not linear.

4.4. Objective and subjective quality assessment of signal samples processed with the AAC codec at different bitrates

Concerning the broad offer of possible radio programs, we decided to expand the range of analyzed material with a group of audio samples divided into 4 categories:

1) musical instruments – castanets, vibraphone, guitar,
2) speech and singing – lector female and male speech, as well as soprano (high female voice), tenor (high male voice) and quartet (soprano, alto, tenor, bass),
3) music genres – a choir and symphony orchestra piece, one electronic and one popular music piece,
4) popular music – well known songs of Michael Jackson, Jamiroquai and Queen, two from each artist.

The test materials from categories 1–3 were sourced from the EBU (*European Broadcast Union*) SQUAM CD (EBU Tech 3253, 2008), whereas the samples from category 4 came from the author's private music library. All reference samples were created as PCM (*Pulse Code Modulation*) WAV files sampled at 48 kHz, 16 bit stereo. The degraded samples were coded at different bitrates using the AAC algorithm. The sampling frequency was also set to 48 kHz. All music files were available for the listeners during training phase. A detailed description of test signals is given in Table 1.

Currently, the digital DAB+ multiplex in Poland offers radio programs transmitted at 6 bitrates: 64, 72, 96, 104, 112 and 128 kbps. For the purpose of this test, the signal samples have been processed using the

Table 1. Audio test signals used in the subjective and objective tests.

Category	File name	Duration [s]	Description
Musical instruments	Castanets	20	Castanets solo
	Vibraphone	15	Vibraphone solo
	Guitar	16	Guitar solo
Speech and singing	Female speech	23	Female lector in English
	Male speech	22	Male lector in English
	Soprano	28	Female singing (higher voice) acapella
	Tenor	29	Male singing (lower voice) acapella
	Quartet	28	Female and male singing (soprano, alto, tenor, basso) acapella
Music genres	Choir	31	Choir with symphonic orchestra
	ABBA	33	ABBA electronic music
	Eddie Rabbitt	21	Guitar with two male singing
Popular music	Billie Jean	27	Popular music piece by Michael Jackson
	Thriller	20	Popular music piece by Michael Jackson
	Little L	24	Popular music piece by Jamiroquai
	Runaway	22	Popular music piece by Jamiroquai
	A Kind of Magic	24	Popular music piece by Queen
	Bohemian Rhapsody	25	Popular music piece by Queen

AAC algorithm at 4 bitrates, with a step of 32 kbps, that is: 64, 96, 128 and 160 kbps.

Tests were carried out in turns, one participant after another, with a short break between each group of audio signals. The signal samples coded at different bitrates were separated by a 0.5–1 s interval. Each individual received the same set of instructions. The actual bitrate of the processed audio material was not mentioned. The results of objective and subjective quality assessment for each category are shown in Figs. 4–7. Obtained subjective results have been treated with the

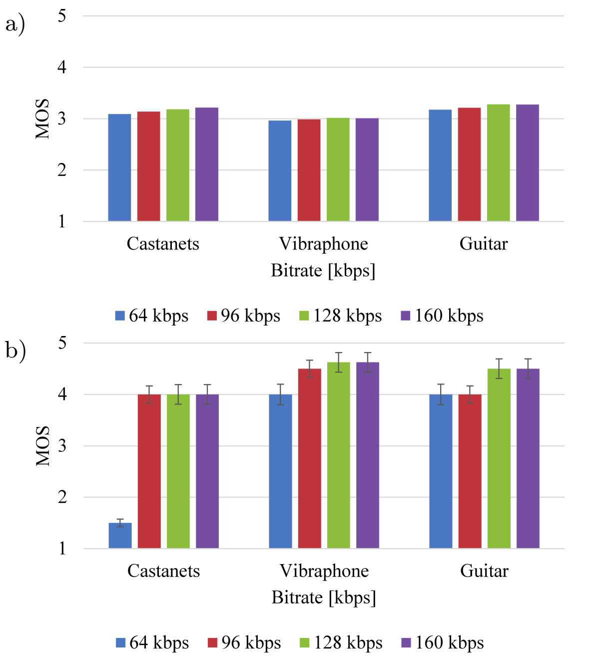

Fig. 4. Musical instruments: a) objective quality assessment, b) subjective quality assessment.

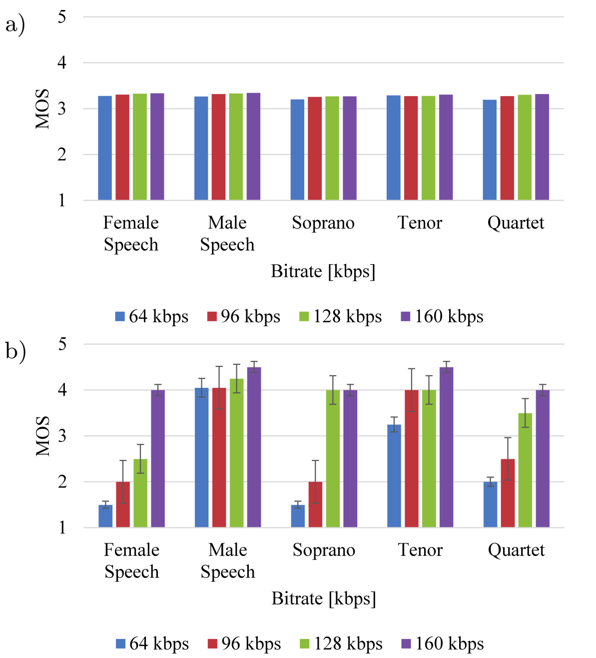

Fig. 5. Speech and singing: a) objective quality assessment, b) subjective quality assessment.

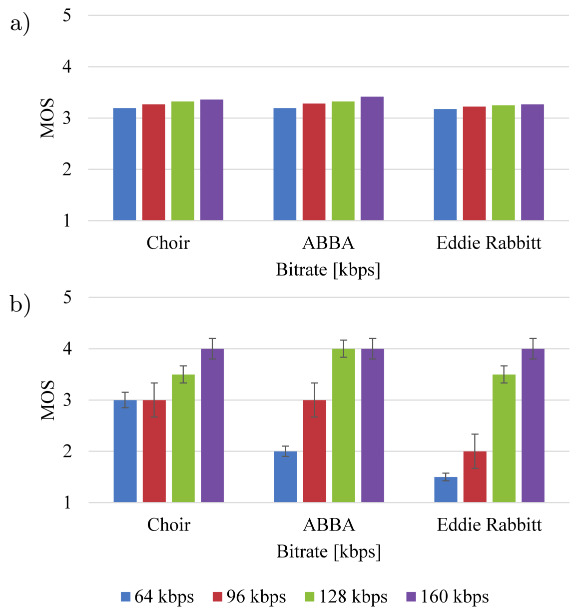

Fig. 6. Different music genres: a) objective quality assessment, b) subjective quality assessment.

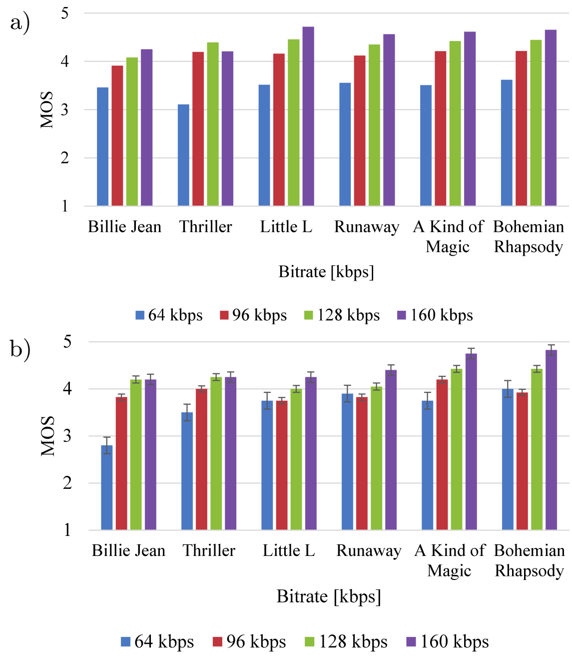

Fig. 7. Popular music: a) objective quality assessment, b) subjective quality assessment.

ANOVA (*Analysis Of Variance*) statistical analysis, as shown in Table 2. The confidence interval was set to 95%.

Table 2. ANOVA test results.

Category	α	P	F_{crit}	F
Musical instruments	0.05	0.16	4.26	2.23
Speech and singing	0.05	0.07	3.06	2.66
Music genres	0.05	0.62	4.26	0.51
Popular music	0.05	0.45	2.77	0.99

According to obtained results, in each case the P value was not less than α. Additionally, the F value did not exceed the F_{crit}. This proves that the hypothesis cannot be rejected.

In case of castanets, significant distortions and artefacts have been noticed, especially for lower bitrates. When considering the clarity of the audio material, both vibraphone and guitar received high grades, regardless of chosen bitrate.

According to the analysis, the AAC codec is much more efficient when it comes to processing audio material containing lower (male speech and tenor) than higher spectrum range (female speech and soprano). It was noticed that lower bitrates led to numerous distortions, perceived as an artificial, metallic and unnatural voice.

In case of music pieces with a clear stereo separation for left and right channel, it is necessary to use higher bitrates. According to the listeners, there was a clearly noticeable effect of a limited scene with a clear cutoff of lower and higher frequencies. This had a significant impact on the overall assessed quality.

The same remarks were given in case of popular music pieces. Bitrates of less than 128 kbps sometimes proved to be insufficient when it comes to providing high-quality audio material.

The objective quality metric proved to be much more accurate when analyzing popular music samples from category 4 than samples from category 1–3. In case of samples from category 4, the software assumption very much resembled scores given by human listeners. On the other hand, the predictions given for samples from category 1–3 were less precise. The score did not vary much, regardless of chosen bitrate.

When it comes to analyzing speech or singing samples, the most important issue is the clarity and transparency of the audio material, since information contained in the voice must reach the listener. According to the subjects, the sound colour was significantly worse for samples coded at lower bitrates, especially 64 kbps. This impression was given regardless of the music genre.

The loss of high frequencies and loss of attack of the transient were also perceived. Furthermore, as the listeners indicated, in case of signal samples from category 3–4, spatial attributes of sound, including spaciousness, sound perspective and localization stability, were reported as annoying or even unacceptable for bitrates lower than 128 kbps. This effect was less common for electronic music pieces, as it was for classical or popular music.

In case of electronic music pieces, some effects such as distortion, unnatural and metallic sound, or even a cutoff of lower or higher frequencies may be viewed by some listeners as intentional.

4.5. Questionnaire

After finishing the listening tests, participants were asked to give answers to two questions in the form of a closed multiple-choice tests, and rank the answers according to their importance from 1 (least important) to 3 (most important). If the digital DAB+ standard is intended to replace the well-known analog FM radio in the nearest future, it seemed quite interesting to learn what people think about this switchover and what would encourage them to migrate from the analog to digital radio domain.

In the first question they were asked what would encourage them, taking into account the ecological, economical and practical aspect, to change or buy a new DAB+ radio receiver. In the second question, they explained which of the switchover criteria they considered as most important. The results of this questionnaire are shown in Figs. 8–9.

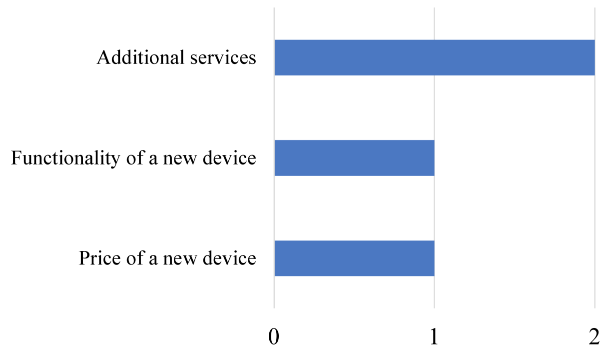

Fig. 8. Factors that would encourage users to buy a new DAB+ radio receiver.

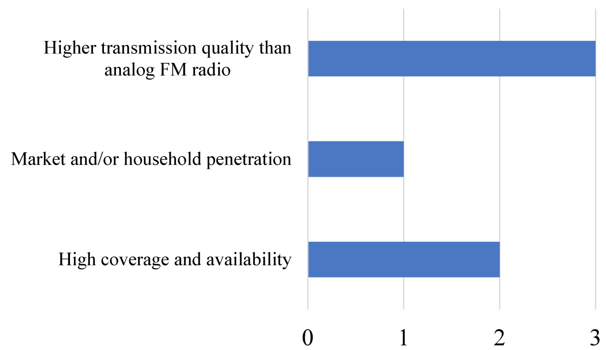

Fig. 9. Most important switchover criteria for migrating from analog to digital radio domain.

According to the participants, the most crucial factor are additional services offered by DAB+. It was quite interesting to notice that aspects such as functionality and price of a new radio receiver came ex aequo at second place with exactly the same number of votes.

Achieving higher transmission quality than analog FM radio was ranked as the most important switchover criteria. The later answers were aspects such as high coverage and availability of the digital radio signal and market and/or household penetration. It is vital to understand that, in order to successfully introduce DAB+, any service needs to offer quality on a superior level to other contemporary services.

5. Summary

According to the study, the DAB+ broadcasting system offers superior quality compared with traditional FM radio transmission. This fact is considered as one of the crucial aspects when it comes to thinking about a nationwide migration from analog to digital radio domain. DAB+ has all the required capabilities to become an efficient replacement for traditional analog FM broadcasting systems.

Of course, analog FM radio, thanks to its widespread and availability, will be used for many more years. Until now, only few countries are considering a total switchover from analog to digital radio domain in the nearest future. This fact should lead to an increase of available mobile and household hybrid FM, DAB+ and Internet radio receivers.

The results show that there is a need for continuous development of objective quality metrics. ViSQOLAudio proved to be a reliable and helpful tool. It can provide valuable feedback during the development and evaluation of any test. However, no objective metric can replace the actual evaluation of user perceived quality.

Furthermore, future studies, e.g. concerning different target groups, are required in order to meet the needs of all listeners. This will inevitably affect the level of user satisfaction. Whenever thinking about development, introduction or maintenance, providing high quality QoS and QoE parameters, especially under limited bandwidth resources, remains a key aspect of any broadcasting system. Of course, efficient bitrate assignment, related with the number of radio programs in a single multiplex, is still one of the most important factors under restricted bandwidth conditions.

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