

3GPP Release 6 Codecs for Audio-Visual Broadcast Service

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Abstract

Broadcast services over wireless networks are gaining popularity with increased network capacity and the emergence of mobile terminals supporting advanced audio and video codecs. Successful deployment of such services relies on efficient bandwidth usage to deliver high quality audiovisual media to the end user. 3GPP Release 6 addresses this issue by standardizing two new advanced audio codecs (AMR-WB+ and E-AAC+) and the new H.264/AVC video codec.

Tests performed with expert listeners and viewers show that video is the limiting factor for overall service quality and that efficient audio compression plays a crucial role. The superior performance of AMR-WB+ at low bitrates and its consistent quality across speech, mixed and music content makes it the preferred choice for broadcast services conveying audiovisual media to a mobile terminal and thereby ensuring best possible service quality for the end user.

Keywords

Codec, 3GPP, broadcast, mobile, audio, video, quality

1. INTRODUCTION

In the recent years, mobile audiovisual media has gained high market focus due to the rapid evolution and development of mobile terminals' multimedia capabilities. Along with this development there has been a tremendous evolution of audio and video data compression technology as well as a steady increase of data transmission capacity in wireless networks. Combined together, these advances have contributed to the demand to support new compelling wireless audio-visual services, such as Mobile TV.

The 3rd Generation Partnership Project (3GPP), which is the standardization body for GSM/UMTS wireless systems, has consequentially taken the action to standardize all required components which enables such services in its system. Corresponding standardization groups of other wireless systems (3GPP2, DVB) have taken similar actions.

To date, the most prominent and mostly talked-about example of mobile broadcast services is Mobile TV,

but there are many other important service examples, such as music and audio book distribution, information services, pod casting, etc. In this paper we mainly focus on the Mobile TV example. However, many of the conclusions are valid for other services that include mobile audio-visual media distribution.

3GPP provides various standardized mechanisms for mobile audio-visual media services. One major distinction between the services is whether the media is distributed using streaming or download. In the former case the media is not stored on the device, but rendered and discarded after reception. Another important distinction is whether the media is distributed according to a unicast (i.e., point-to-point) or a multicast (point-to-multipoint) mechanism. Unicast services in 3GPP are realized as, e.g., transparent end-to-end Packet-switched Streaming Service (PSS) [1] or as Multimedia Messaging Service (MMS) [2] and existed also in earlier 3GPP releases prior to Release 6. These are not broadcast services although their end-user experience may be very similar to that of a corresponding broadcast service. In fact, during early service-deployment phases with correspondingly low penetration, it is even more efficient to realize broadcast audio-visual services by means of unicast rather than multicast mechanisms. However, in order to cope with high service-penetration cases, 3GPP Release 6 provides a true broadcast service, the Multimedia Broadcast and Multicast Service (MBMS) [3], that is more resource efficient than unicast.

In order to deal with increasing quality while maintaining bit rate efficiency demands for mobile audio-visual services, 3GPP Release 6 also specifies new and highly efficient audio and video codecs, which represent the latest available compression technology. In this paper we investigate how much these new codecs can improve mobile audio-visual services over those services that are based on the codecs available in the earlier 3GPP releases.

This paper is organized as follows. First we describe in more detail how broadcast/multicast services are realized in 3GPP and we also highlight certain commonalities with corresponding services in other cellular and non-cellular systems, such as 3GPP2 and

DVB-H. Then we provide an overview of audio and video codecs available in 3GPP. In the following section we provide the results of a quality assessment of these codecs in typical audio-visual service applications. Finally, we conclude the paper with a summary of the most important findings.

2. BROADCAST IN 3GPP SYSTEMS

During 2003, 3GPP and 3GPP2 started to address broadcast/multicast services in GSM/WCDMA and CDMA2000, respectively. In 3GPP the work item is called Multimedia Broadcast and Multicast Service (MBMS). In 3GPP2 it is called BroadCast and MultiCast Service (BCMCS). The specifications of cellular broadcast services were functionally frozen during 2004. 3GPP MBMS and 3GPP2 BCMCS have many commonalities.

Whereas MBMS and BCMCS introduce efficient support for broadcast/multicast transport in cellular networks, OMA BCAST is working on the specification of broadcast/multicast related service layer functionalities, which can be applied not only to cellular but also to non-cellular digital broadcast networks. OMA BCAST addresses for instance content protection, service and program guides, transmission scheduling and so on. OMA BCAST is agnostic of the underlying broadcast/multicast distribution scheme, which could be MBMS, BCMCS or a non-cellular digital broadcasting system like DVB-H.

Both MBMS and BCMCS are introducing only small changes to the existing radio and core network protocols. The same applies to OMA BCAST for broadcast/multicast specific service layer functions.

MBMS and BCMCS add the following features to cellular networks:

- A set of functions that control the broadcast/multicast delivery service; MBMS uses the term Broadcast/Multicast Service Center whereas in BCMCS it is called "BCMCS Controller"

- Broadcast/multicast routing of data flows in the core network
- Efficient radio bearers for point-to-multipoint radio transmission within a cell
- In addition MBMS and BCMCS also specify protocols and media codecs for the delivery of multimedia data. However, many of those protocols and all media codecs are not a new feature but are "shared" with other services like on-demand unicast streaming.

Figure 1 indicates which nodes of the 3GPP architecture that are affected by MBMS. It also highlights the new BM-SC function, which is responsible for providing and delivering cellular broadcast services. It serves as an entry point for content-delivery services that want to use MBMS. Towards the mobile core network it sets up and controls MBMS transport bearers and it can be used to schedule and deliver MBMS transmissions. The BM-SC also provides service announcements to end-devices. These announcements contain all necessary information - such as multicast service identifier, IP multicast addresses, time of transmission, media descriptions, and so on - that a terminal needs in order to join an MBMS service. The BM-SC can be used to generate charging records for data transmitted from the content provider. It also manages the security functions that 3GPP has specified for the multicast mode.

In addition to the streaming delivery method supported by both MBMS and BCMCS, MBMS also supports download. MBMS download can be used to deliver arbitrary files from one source to many receivers efficiently. Existing content-to-person MMS services, for instance a service that delivers short video clips of a sports event via MMS, will greatly benefit from this feature. Today those services use point-to-point connections for MMS delivery with all the drawbacks mentioned before. In the future the existing MMS sub-system can be easily interfaced with a BM-SC which then distributes the clip via MBMS download. In contrast to MBMS, BCMCS does not define protocols that explicitly support the file delivery service.

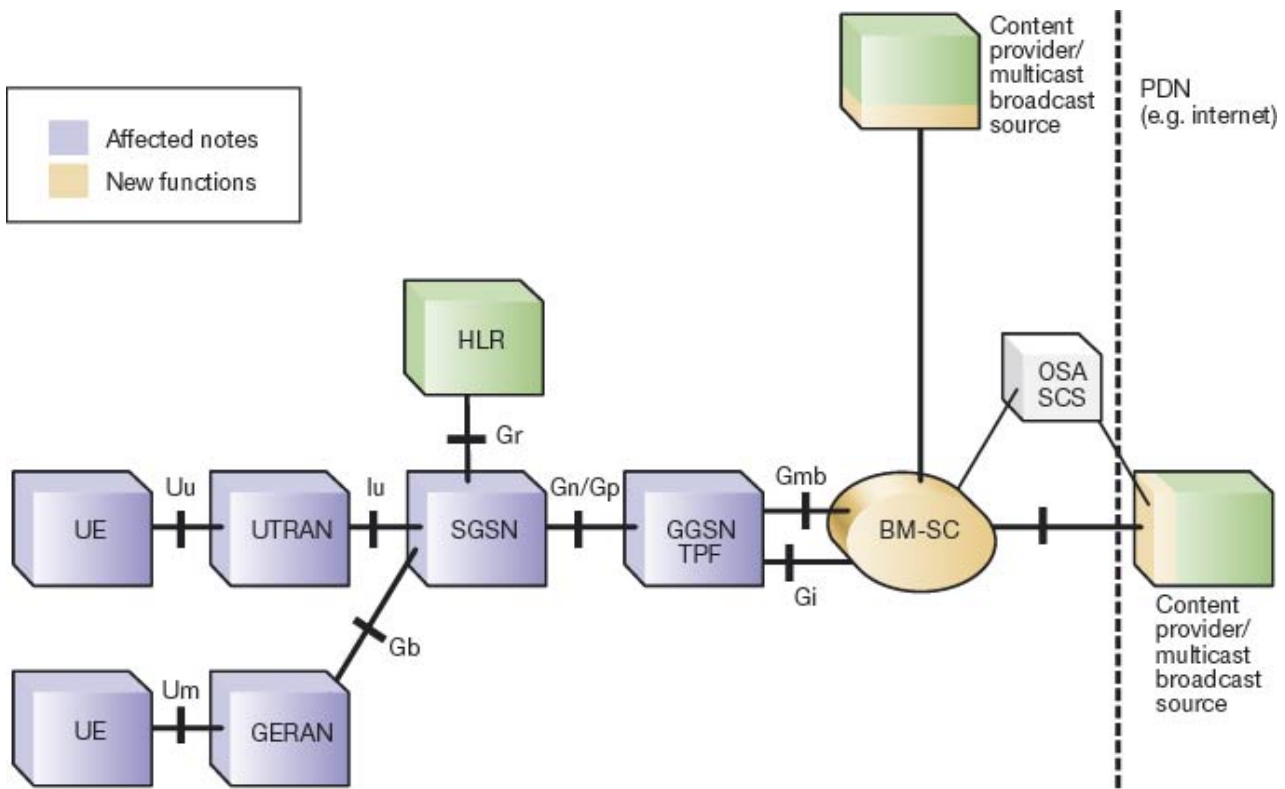


Figure 1: 3GPP nodes affected by MBMS. The basic 3GPP2 architecture differs somewhat from that of 3GPP, but has been amended in a similar manner to accommodate BCMCS

MBMS in UTRAN offers up to 256 kbps per single MBMS channel for services. An UTRAN cell may offer up to 800 kbps for MBMS services. This assumes that terminals on cell borders receive radio signals from more than one radio base station and uses soft combining. If the receiving terminals support receiver diversity, the capacity of MBMS cells can be almost doubled.

MBMS in GSM/EDGE Radio access network (GERAN) may use up to 4 Time Slots in downlink for a single MBMS Channel. Depending on the modulation scheme and the network dimensioning, a channel capacity between 32kbps and 128kbps can be achieved. The total cell capacity depends on the number of supported frequencies of that cell.

3. 3GPP AUDIO AND VIDEO CODEC STANDARD OVERVIEW

3.1 3GPP Audio Codecs

3GPP PSS, MMS and MBMS specifications define Speech and Audio media types, which can both be used for the audio part of the audiovisual media. The Speech media type, if supported, mandates the use of AMR [4] or AMR-WB [5] speech decoders depending on if the client supports narrow band or wideband

speech. If Audio media type is supported, then earlier than 3GPP Release 6 specifications of PSS and MMS recommend the use of the MPEG-4 AAC Low Complexity (AAC-LC) object type decoder. The distinction between Speech and Audio media types was necessary since no single codec existed equally well coping with both of them.

This situation and the fact that AAC-LC did not provide sufficient quality at low bit rates is the background of an audio codec standardization effort in 3GPP Release 6 which was seeking for an audio codec providing high quality at low bit rates as well as consistent quality over speech and music audio content. This led in turn to the standardization of two new audio codecs for the Audio media type, namely AMR-WB+ [6] and Enhanced AAC+ [7], which both are recommended for 3GPP Release 6 PSS, MMS, and MBMS services.

These newly standardized Release 6 codecs have similar bit-rate operating ranges but perform differently depending on the bit rate and the content type. 3GPP has characterized the performance and other attributes of both codecs and a detailed report is available under [8].

Both audio codecs support mono and stereo coding as well as sampling rates up to 48 kHz. On the

algorithmic level, the two codecs are quite different. While Enhanced AAC+ follows the traditional audio coding principles by using a transform coding, AMR-WB+ uses a breakthrough hybrid coding model, incorporating both an ACELP speech coding scheme and a transform based audio coding scheme. This allows it to cope both with generic audio content as well as standard speech and music content. The use of parametric stereo coding principles as well as bandwidth extension is adopted by both codecs in order to reduce the amount of bits necessary for high quality coding. More detailed information of AMR-WB+ can be found in [9].

3.2 3GPP Video Codecs

When MBMS was finalized for Release 6, 3GPP took advantage of the latest video technology standardized by ITU-T and MPEG. Consequently, ITU-T Recommendation H.264, also known as MPEG-4 AVC [10], is the video codec recommended for MBMS. Other 3GPP services (PSS and MMS) also support H.263 Baseline, H.263 Profile 3 and MPEG-4 Visual.

The improved coding efficiency of H.264/AVC comes with the price of complexity, which is a concern for constrained terminals such as mobile devices. However, since decoding H.264/AVC is not as demanding as encoding it, the challenge for an MBMS terminal decoding H.264/AVC is still within reason. Compared to H.263, decoding H.264/AVC is roughly 2-3 times more complex, to a large extent due to the advanced motion compensation and the enhanced de-blocking filter

For MBMS in Release 6, the H.264/AVC Baseline profile, or more precisely, its Main subset, and Level 1b are used: video bit rates up to 128 kbps are supported and the picture size can be up to QCIF (176x144 pixels) at 15 frames per second.

Although H.263 is not recommended for MBMS, it may actually be used. The reason is that H.263 Baseline is the required video codec for the Packet-Switched Streaming Service (PSS) as well as all other 3GPP services supporting video. Given its unique status in 3GPP, it may be used for MBMS as well.

4. PERFORMANCE EVALUATIONS

4.1 3GPP/MPEG official evaluations

4.1.1 3GPP Audio Codecs

Speech and audio codecs suitable for 3GPP audiovisual services have been evaluated in various quality assessments. The most comprehensive evaluations are found in the respective 3GPP codec

characterization technical reports. These are 3GPP TR 26.975 and TR 26.976 for AMR and AMR-WB speech codecs, respectively, and TR 26.936 for 3GPP Rel-6 AMR-WB+ and E-AAC+ audio codecs. In addition TR 26.936 also contains a quality comparison for generic audio content with AMR-WB and MPEG AAC-LC which were used as performance references during the Release-6 audio codec selection. The Release-6 audio codec selection even considered MPEG-4 HE AAC,v1 (aacPlus) as a further codec candidate, which, however was discarded due to insufficient quality and its failure to meet the defined minimum performance requirements.

While all details on the various evaluations of the standardized 3GPP speech and audio codecs available for 3GPP audio-visual services are found in the respective characterization TRs, a short summary is as follows:

1. AMR and AMR-WB are speech codecs and most suitable for speech content only. Both codecs are designed and optimized for conversational speech services in cellular networks. Accordingly, they have a low coding delay of only 25 ms and introduce a frequency band limitation to the narrowband telephony band (about 100-3500 Hz) in case of AMR, and the wideband telephony band (about 60 – 7000 Hz) in case of AMR-WB, respectively.
2. MPEG-4 AAC and MPEG-4 HE AAC,v1 (aacPlus) are audio codecs that are less suitable for low bit rates required for the various audiovisual services defined in 3GPP mobile systems. At low rates (below around 32 kbps) these codes have more or less severe problems especially with speech content and in stereo operation.
3. In total, on general audio content including music, speech and mixed material, AMR-WB+ and E-AAC+ perform substantially better than any other of the mentioned speech and audio codecs.
4. In a direct comparison between AMR-WB+ and E-AAC+, the following picture is obtained:
 - a. For speech, AMR-WB+ is always better than E-AAC+.
 - b. For mixed content (both speech and music), AMR-WB+ is generally better than E-AAC+.
 - c. For monophonic operation and a range of bit rates from 10 – 32 kbps, AMR-WB+ is better than E-AAC+. In mono E-AAC+ reduces to HE AAC,v1 which was disqualified during the 3GPP audio codec selection.

- d. For high-bit rate stereo music, E-AAC+ is better than AMR-WB+.

4.1.2 3GPP Video Codecs

In contrast to the case of speech and audio codecs, 3GPP has so far not seen the need for performing official video codec characterizations. All video codecs adopted by 3GPP target the same applications and content types, whereas the main differences lie in efficiency and complexity. The codecs have been standardized by ITU-T and/or MPEG and there exist a large number of comparative tests between these in the literature. Nevertheless, during the Release 6 video codec selection process, 3GPP made strict requirements in terms of quality gains over existing codecs. H.264/AVC qualified and was in the end the only candidate submitted for

and video device capabilities, user expectation, etc. While this means that it is very difficult, if not impossible, to derive a valid service quality judgment for all conceivable cases, our attempt was to obtain an informal judgment under the following assumptions and parameter settings. As audio-visual codec combinations we consider H.263/AMR (both mandatory) and MPEG-4Visual/MPEG-4AAC from 3GPP Release 5, and the combinations H.264/AMR-WB+ and H.264/E-AAC+ from Release 6. The chosen video format was QCIF (176x144 pixels), whereas the video frame rate varied depending on content and bit rate. Audio was encoded in mono or stereo depending on the content, the chosen audio bit rate and the capabilities of the codec. The considered gross bit rates (for transport) were 40, 64 and 128 kbps, which due to IP overhead results in somewhat

Bearer	Release 5 codecs		Release 6 codecs
	H.263 / AMR (default codecs)	MPEG-4 / AAC	
40 kbps	<i>Unacceptable</i>	<i>Unacceptable</i>	<i>Acceptable</i> depending on content and use of AMR-WB+
64 kbps	<i>Unacceptable</i> due to video	<i>Unacceptable</i>	<i>Useful</i> depending on content and use of AMR-WB+
128 kbps	<i>Limited</i> due to audio	<i>Useful</i>	<i>Useful</i> no gain can be expected for audio bit rates above 24 kbps

Table 1: Overall service quality enabled with Rel-5 and Rel-6 codec

selection.

In the context of video performance it should be mentioned that 3GPP has started a new work item in Release 7 on video codec performance requirements for all packet-switched multimedia services including MBMS. The expected outcome is an informative technical report detailing test conditions and metrics for “self certification” of codec implementations. Considering that video decoders (but not encoders) are standardized and that error concealment is implementation specific, there is room for vendor differentiation when it comes to end-user quality.

4.2 Service quality

The quality experience of an audio-visual service is a combination of various constituents such as audio quality, video quality, audio-video synchronicity, audio

lower net media bit rates of 36, 56 and, 115 kbps, respectively. The media bit rate division between video and audio bit rate was adjusted for each codec combination in order to get the best possible overall audiovisual quality for it. For the two Release 6 codec combinations the same bit rate divisions were used.

4.2.1 Assessment methodology

The quality assessments were made based on a broad selection of audiovisual content that can be regarded “typical” for audiovisual service applications such as Mobile TV or other audiovisual streaming services. Specifically, audiovisual clips of about 1 min duration each were considered from the following categories: Animation, cartoon, football, movie, music, news, and soap opera. The assessment was done based on expert quality judgments from various experts, who were asked to grade each clip encoded

Codexes Bitrates (kbps)	H.263/AMR		MPEG-4/AAC		H.264 baseline/ AMRWB+		H.264 baseline/ E-AAC+	
	Video	Audio	Video	Audio	Video	Audio	Video	Audio
Media: 36 Transport: 40	Video	Audio	Video	Audio	Video	Audio	Video	Audio
	Combined AV		Combined AV		Combined AV		Combined AV	
Media: 56 Transport: 64	Video	Audio	Video	Audio	Video	Audio	Video	Audio
	Combined AV		Combined AV		Combined AV		Combined AV	
Media: 115 Transport: 128	Audio	video	Video	Audio	Video	Audio	Video	Audio
	Combined AV		Combined AV		Combined AV		Combined AV	

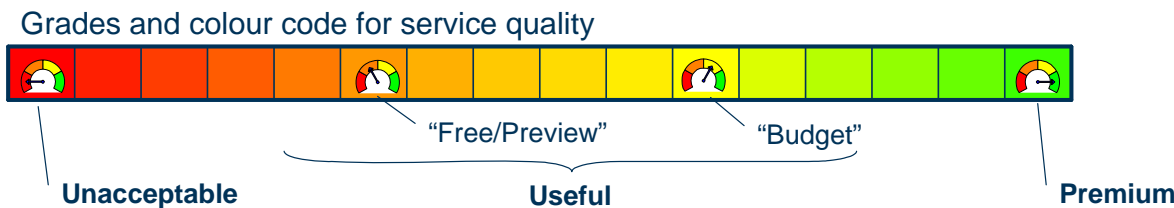


Figure 2: Average Performance across Content

with the above mentioned audiovisual codec combinations and at the given media bit rates. The experts were first asked to give separate judgments for the video and the audio quality. Under the assumption that a quality limitation of either the video or the audio will limit the overall quality experience of the service, the service grade is calculated as the minimum of individual grades for video and audio. In a final step the grades were polled across the different audiovisual clips in order to get average performance figures for the different codec combinations and considered bit rates.

The defined grading categories for video and audio separately and the combined service grade were

- Unacceptable: Never wanted, not even for a free service
- Useful: with sub-categories
 - Free/Preview: Acceptable if no service charge
 - Budget: Acceptable for low service charge
- Premium: More than sufficient quality for a regularly charged service

4.2.2 Service Quality Conclusions

An overview of the audiovisual service quality obtained with the various considered codec

combination is provided in Figure 2 and conclusions are summarized in Table 1.

In general, it is found that the video quality is limiting overall service quality. An immediate implication is that most of the bit rate needs to be assigned to the video whereas the audio bit rate should be as low as possible. The Release 6 audiovisual codecs provide a substantial quality improvement over any Release 5 codec combination. It is found that they generally provide useful service quality on a 128 kbps bearer. On a 64 kbps bearer useful service quality is enabled depending on content and use of the AMR-WB+ audio codec. Even on a 40 kbps bearer acceptable service quality is enabled depending on content and provided that AMR-WB+ is used for the audio. The Release 5 codecs cannot generally provide acceptable service quality on a bearer with less than 128 kbps.

Services based on H.263/AMR on a 64 kbps bearer and below suffer largely from insufficient video quality of H.263. On a 128 kbps bearer H.263 performance is becoming acceptable for most content, however, now the limited audio quality of the AMR speech codec causes the overall service quality limitations.

Services based on MPEG-4Visual/AAC on a 64kbps bearer and below suffer mainly from poor AAC audio quality at the involved low audio bit rates. On a 128

kbps bearer an overall quality suitable for useful services can be achieved.

A direct comparison of the audio quality enabled with the Release 6 audio codecs shows that AMR-WB+ is clearly preferable over E-AAC+ for 40 and 64 kbps bearers. It allows premium audio quality already on a 64 kbps bearer and budget quality on a 40 kbps bearer. A service using E-AAC+ suffers with significantly higher likelihood from poor audio quality on a 64 kbps bearer. No acceptable services with E-AAC+ are possible on a 40 kbps bearer. On a 128 kbps bearer, due to video quality limiting the overall service quality, no real gain for overall service quality can be expected with audio bit rates higher than 24 kbps.

5. CONCLUSIONS

In this paper we review the recent standardization efforts of mobile broadcast solutions in 3GPP and in particular the audiovisual quality that can be obtained for different usage scenarios. 3GPP Release 6 introduces the Multimedia Broadcast and Multicast Service (MBMS) together with advanced audio (AMR-WB+ and E-AAC+) and video (H.264/AVC) codecs that provide significant service improvements at lower bit rates.

In order to assess the performance of these codecs for services with limited radio bandwidths, we have conducted a number of combined expert listening and viewing tests. These tests show that an acceptable service quality for mobile TV may be realized on radio bearers of 64 kbps, or even with 40 kbps bearers, depending on content.

Video requires the largest part of the available bit rate and is therefore the limiting factor for the overall end-user quality. With larger display resolutions than the current 176x144 pixels supported in Release 6, the demand for higher video bit rates will increase at the expense of audio bit rate. In order to maintain a high service quality it is therefore necessary to focus on efficient audio coding at low bitrates. Of the two 3GPP Release 6 codecs, AMR-WB+ is preferable due to its better performance at low bit rates and its more consistent quality across various content types.

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