Technical Specification Group Services and System Aspects **TSGS#19(03)0164**
Meeting #19, Birmingham, UK, 17-20 March 2003

Title:	WID for Higher Bitrate Audio Codec
Source:	Dolby Laboratories, Apple Computer, AT&T Wireless Services, RealNetworks
Agenda Item:	7.4.3
Document for:	Approval

Work Item Description

Title

Higher Bitrate Audio Codec for packet-switched streaming and messaging services

1 3GPP Work Area

	Radio Access
	Core Network
X	Services

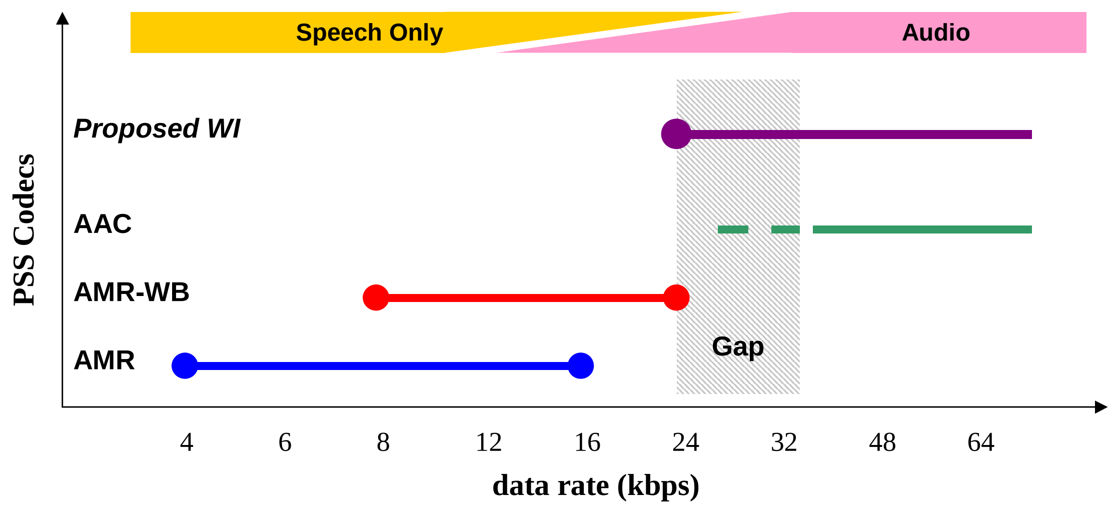
2 Linked work items

TSG-SA WG4 Extended packet switched streaming service (Rel-6)

3 Justification

The 3GPP packet switched streaming service currently covers streaming speech and audio with AMR, AMR-WB and MPEG-4 AAC codecs. This set of codecs provides useful quality for wideband speech and narrowband speech at low to very low data rates (<24 kbps), and general audio at moderate and higher data rates (>32 kbps/channel).

It has been recognized within SA4 that this leaves a gap in the 24 to 32 kbps/chan data rates for general audio and for high quality speech at data rates above 24 kbps. Above 24 kpbs AMR-WB does not significantly improve in quality for speech or audio. Below 32 kbps AAC speech quality deteriorates, and audio bandwidth becomes severely restricted.



It has also been recognized that for many audio services, especially when accompanied by video, the audio data rate will need to be as low as possible, including the sub-32 kbps range.

Given the data rate gap between the currently specified codecs, and the importance of delivering content at such data rates, it is clear that a new codec that would eliminate the gap is highly desirable. In response this need, SA4 has arrived at the following working assumption for audio codecs for streaming and messaging services:

- A codec for audio in the lower bitrate audio range (currently specified as <=32kbps) shall be mandated.
- A codec for audio in the higher bitrate audio range (currently specified as >=24kbps) shall be mandated.

4 Objective

Standardisation of a Higher Bitrate Audio Codec for packet-switched streaming and messaging services targeting Release 6.

The objectives of the Higher Bitrate Audio Codec include the following:

- High audio quality for music, mixed content and speech items at data rates as low as 24kbps.
- Capable of “CD quality” audio reproduction at higher data rates.
- Mono and stereo encoding modes.
- Audio quality as good as AAC and exceeding AAC at one or more data rates in one or more usuage modes.
- Able to decode MPEG-4 AAC LC bitstreams.

The resulting codec will be considered as a candidate for the Higher Bitrate Audio Codec (>=24kbps) for PSS and MMS services in 3GPP Rel-6.

5 Service Aspects

The work item does not introduce any new services. It extends the currently recommended AAC codec for use in packet-switched streaming and messaging services. The codec is primarily intended for non-conversational services. As this work item extends current recommended codec, it is expected that there are no service or architectural impacts. SA1 and SA2 will be kept informed of the progress of the work in SA4.

6 MMI-Aspects

None

7 Charging Aspects

Outside the scope of this work item. Covered in the linked PSS and MMS service work.

8 Security Aspects

Outside the scope of this work item. Covered in the linked PSS and MMS service work.

9 Impacts

Affects:	USIM	ME	AN	CN	Others
Yes		X			
No	X		X	X	
Don't know					

10 Expected Output and Time scale (to be updated at each plenary)

New specifications						
Spec No.	Title	Prime rsp. WG	2ndary rsp. WG(s)	Presented for information at plenary#	Approved at plenary#	Comments
	New audio codec specification					

Affected existing specifications				
Spec No.	CR	Subject	Approved at plenary#	Comments
			SA#21	

11 Work item raporteurs

Charles Robinson, Dolby Laboratories Inc.

12 Work item leadership

TSG-SA WG4

13 Supporting Companies

Dolby Laboratories, Apple Computer, AT&T Wireless Services, RealNetworks

14 Classification of the WI (if known)

	Feature (go to 14a)
X	Building Block (go to 14b)
	Work Task (go to 14c)

14a The WI is a Feature: List of building blocks under this feature

(list of Work Items identified as building blocks)

14b The WI is a Building Block: parent Feature

PSS Rel-6

14c The WI is a Work Task: parent Building Block

(one Work Item identified as a building block)

TSG-SA4#25 meeting
January 20-24, 2003, San Francisco, CA, USA

Tdoc S4 (03)0014

Source: **Dolby Laboratories Inc.**

Title: **Higher Bitrate Audio Codec Description**

Document for: **Information**

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1. Introduction

This document introduces a new audio codec to be submitted for consideration under the Higher Bitrate Audio Codec WI.

As indicated in the working draft of 3GPP TS 26.234, SA4 PSM has agreed that specification of mandatory audio codec(s) is desirable for Release 6 [1].

> SA4 PSM was in agreement that the selection of a mandatory codec for audio in PSS and MMS (and MBMS ffs) would be desirable in the context of Rel.6. The group acknowledged that in the lower bitrate audio range (12 kbit/s to <32 kbit/s, as defined in the S4-020660) there were two contenders being presented, namely aacPlus and the proposed Wideband AMR Extension presented as a work item to SA4. In the higher bitrate audio range, the group agreed that at the present moment, aacPlus and AAC appear to be the contenders in that field.

Thus, two classes of codec have been declared, currently differentiated as "lower bitrate audio," and "higher bitrate audio."

A work item to specify a lower bitrate audio codec (SP-020686) has been created with AMR-WB+ as the only currently declared candidate. A work item to specify a higher bitrate audio codec has been proposed. This document describes a codec that shall be a candidate under the Higher Bitrate Audio Codec WID.

2. References

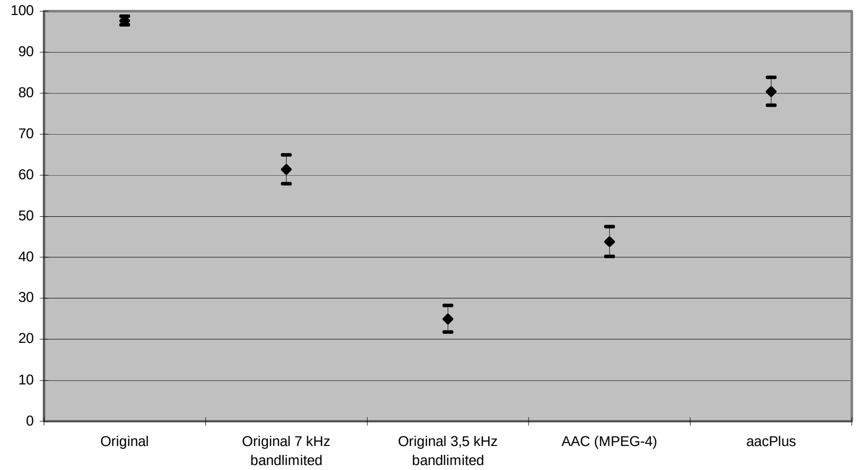
[1] Tdoc S4-030149, Working Draft 3GPP TS 26.234: "Transparent end-to-end packet switched streaming service (PSS); 3GPP file format (3GP)".

3. Technical Brief

The new codec builds on AAC technology to provide effective audio coding at lower data rates. It is compatible with the MPEG-2 and MPEG-4 AAC LC profile standard. In particular, the new decoder can decode MPEG standard AAC LC bitstreams. Furthermore, a standard AAC decoder can decode a bitstream generated by the new encoder and produce a (reduced resolution) audio signal.
The new codec is effective at data rates as low as 14 kbps for mono signals, and 28 kbps for stereo signals. At these low data rates, internal listening tests show that the new codec is competitive with all other audio codecs currently available. At higher data rates, the new codec reverts to standard AAC encoding and decoding and provides the high sound quality for which AAC is already recognized. Unlike AAC, the new codec provides a full bandwidth signal at all data rates.

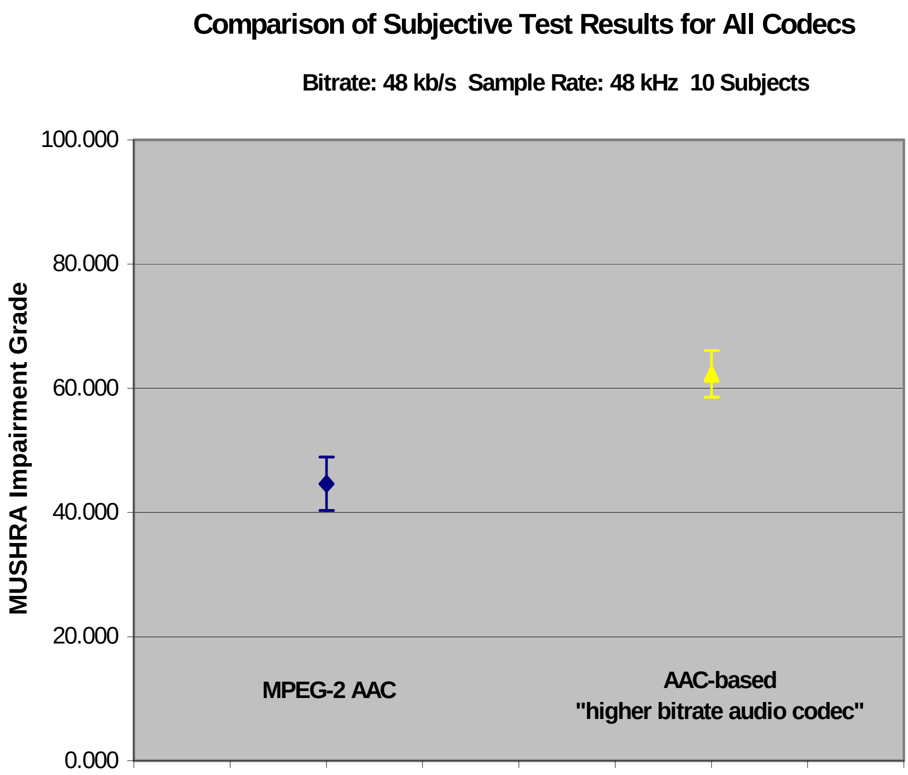
4. Listening test results

As indicated in the WID for Higher Bitrate Audio Codec [3], and as reasonably required to replace the current recommended codec, the new codec must provide



> "Audio quality as good as AAC and exceeding AAC at one or more data rates in one or more usage modes."

A listening test was conducted by the authors using the MUSHRA test method for differentiating audio impairments. The test was conducted with 10 experienced listeners. The test items consist of common "codec killer" test items, e.g. English Speech, German Male Speech, Bagpipes, Harpsichord, etc. The items were coded at 48kbps stereo. The compiled listener grades are displayed in the table below.



The results show that the proposed High Bitrate Audio Codec is significantly better than AAC, in compliance with SA-4 and WID design constraints.

TSG-SA WG4#25bis meeting
February 24 – 28, 2003, Berlin, Germany

Tdoc S4 (03)0262

Source: Dolby

Title: Updated draft proposal for joint Higher Bitrate Audio Codec and PSM audio codec testing for PSS/MMS, version 0.3

Document for: Discussion and approval

Agenda Item:

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1. Introduction

According to the Higher Bitrate Audio Codec WID [1], the resulting codec will be considered as a candidate for PSS and MMS services in 3GPP Rel-6. SA4 will define the selection criteria for a codec for high-rate high quality audio applications. To proceed with the work efficiently, this document proposes a joint audio codec selection testing process for high-rate high quality audio streaming and messaging in SA4.

2. Joint Higher Bitrate Audio Codec and PSM audio codec testing for PSS/MMS

It is proposed to combine the testing effort in selection of the Higher Bitrate Audio Codec and the audio codec for higher bit rate PSS/MMS services. Since the performance requirements of the Higher Bitrate Audio Codec are very much aligned with the expected requirements of higher bit rate audio coding in PSS/MMS, the joint testing seems to be feasible. In addition, combined testing would save resources and enable thorough testing.

The joint testing process could begin after the TSG-SA4#26 meeting in May 2003 according to the time plan [2]. The results should be available in TSG-SA4#27 meeting in July 2003 for codec selection.

3. References

[1] S4-030193 "Work Item Description for Higher Bitrate Audio Codec"
[2] S4-030098 "PSS R6 Time Plan version 1.0"

TSG-SA WG4#25bis meeting
February 24 – 28, 2003, Berlin, Germany

Tdoc S4 (03)00256

Source: Editor (Dolby)

Title: Higher Bitrate Audio Codec Development Schedule

Version 0.1

1. Introduction

This document provides the development schedule associated with the Work Item Description for Higher Bitrate Audio Codec. This codec is proposed as a candidate for packet switched streaming services (PSS) and messaging services (MMS).

2. Development Schedule

The table below gives a draft schedule for development of the higher bit rate codec. The schedule assumes this document and the WID are approved at SA4#25bis.

Date	Action	SA4 meetings	TSG-SA meetings
February	High Bitrate Audio Codec WID approved by SA4 List of proposed codecs for PSS (and MMS) closed	SA4#25bis (24 - 28 Feb)	
March	High Bitrate Audio Codec WID presented for approval at TSG-SA#19.		TSG-SA#19 (17-20 March)
April			
May	Higher Bitrate Audio Codec Design Constraints, Performance Requirements, and Schedule approved. PSS/MMS Audio Codec Selection criteria available from PSM SWG. PSS/MMS Audio Codec Selection Test Plan available from PSM SWG. PSS/MMS Audio codec selection testing starts. (Including some joint experiments with Higher Bitrate Audio Codec selection tests)	SA4#26 (5 - 9 May)	
June	PSS/MMS Audio codec testing on-going		TSG-SA#20 (9-12 June)
July	Selection of Higher Bit Rate Audio Codec (completion of WI). Selection of “high bit rate” audio codec for PSS/MMS.	SA4#27 (7 - 11 Jul)	
August			
September	PSS/MMS Audio codec Test Results available. Review of the PSS/MMS Audio codec test results. Select Audio codecs (low and high bit-rates) for PSS/MMS Rel-6. Draft codec specifications available (prepared beforehand by each candidate). Draft PSS/MMS specs (TS 26.234 and TS 26.140),	SA4#28 (1-5 Sept)	

	updated to specify selected audio codecs, presented for information. Presentation of PSS/MMS Audio codec draft specs for information at TSG-SA#21. Draft PSS/MMs specs (TS 26.234 and TS 26.140) presented for information at TSG-SA#21.		TSG-SA#21 (22-25 Sept)
October			
November	CRs to PSS/MMs specs (TS 26.234 and TS 26.140) presented for approval at TSG-SA4#29.	SA4#29 (24-28 Nov)	
December	Presentation of PSS/MMS Audio codec specs for approval at TSG-SA#22. CRs to PSS/MMs specs (TS 26.234 and TS 26.140) presented for approval at TSG-SA#22.		TSG-SA#22 (15-18 Dec)

6. Conclusion

This schedule is in alignment with current Rel 6 schedule including finalisation of candidate codec list, codec testing and codec selection.

TSG-SA WG4#25bis meeting
February 24 – 28, 2003, Berlin, Germany

Tdoc S4 (03)0254

Source: Editor (Dolby)
Title: Draft Higher Bitrate Audio Codec Performance Requirements

Version: 0.1

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1. Introduction

This document provides the performance requirements associated with the Work Item Description for Higher Bitrate Audio Codec. This codec is proposed as a candidate for packet switched streaming services (PSS) and messaging services (MMS).

2. Performance requirements

Unless otherwise stated, the performance requirements and objectives shall be interpreted as "not worse than" the performance of the reference codec. Conditions "not worse than", "equivalent" and "better than" shall be determined statistically at the 95% confidence interval.

The table below lists the performance requirements against which the codecs complying with the design constraints will be evaluated in terms of audio quality and error robustness.

Criteria	**Performance Requirement**
Audio quality	Better than AAC in at least one test case and equivalent or better in the others. Test cases are 24kbit/s mono, 24kbit/s stereo and 48kbit/s stereo. The codec shall offer the ability to create CD-quality streams at no more than 128kbit/s
Error Robustness	Under 1% frame loss rate (defined below) and at 24kbit/s stereo, candidate shall perform better than error-free AAC-LC at 24kbit/s stereo.

3. Content Types

Pop – with or without vocals
Classical – with or without vocals
Single Instruments
Vocal
Choir
Mixed speech and Music
Clean Speech

4. General considerations regarding audio quality assessments

In tests conducted over the past years, the MUSHRA method (as described in [1]) has been proven to deliver reliable results for the purpose of assessing the audio quality of lossy codecs, especially in the intermediate quality range. Therefore, test data considered in the selection process should be based on the MUSHRA test methodology.

For the purposes of minimizing the workload whilst maintaining integrity, the selection process should be based on relevant test data, previously generated inside 3GPP or by other reputable independent organizations. Minor deviations from the originally intended test scenario are acceptable in this context.

1% random frame loss represents the loss of one frame in 100 under random distribution conditions.

5. References

[1] ITU, *Method for the subjective assessment of intermediate quality level of coding systems,* Recommendation ITU-R BS.1534

TSG-SA WG4#25bis meeting
February 24 – 28, 2003, Berlin, Germany

Tdoc S4 (03)00xxx

Source: Editor (Dolby)

Title: Higher Bitrate Audio Codec Design Constraints

Version 0.1

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1. Introduction

This document provides the design constraints associated with the Work Item Description for Higher Bitrate Audio Codec. This codec is proposed as a candidate for packet switched streaming services (PSS) and messaging services (MMS).

2. Design constraints

The Higher Bitrate Audio Codec is designed for audio delivery applications in both packet switched streaming services and messaging services. In the case of packet switched streaming services, the client would need only feature the Higher Bitrate Audio Codec decoder as audio content will be received solely from a centralised delivery service. For messaging services it is necessary to implement both the High Bitrate Audio Codec encoder and decoder in the streaming client, and as such the encoder must be implementable on current handset platforms.

Design constraints		Notes
Complexity requirements[i]	WMOPS and RAM requirements are given for mono encoder/decoder. For stereo coding, two times the wMOPS and RAM of the mono encoder/decoder is allowed. ROM requirements are independent on the number of channels.	
Encoder	MIPs ≤ 2 x AAC encoder RAM ≤ 4 x AAC encoder ROM ≤ 2 x AAC encoder Program ROM ≤ 2 x AAC encoder	
Decoder	MIPs ≤ 2 * AAC decoder RAM ≤ 4 x AAC decoder ROM ≤ 2 x AAC decoder Program ROM ≤ 2 x AAC decoder	
Algorithmic delay	The algorithmic delay shall not be larger than 200ms.	
Input sampling rate and bit rates	The codec will operate on 32, 44.1 and 48 kHz input sampling rates. The maximum allowed bit rate shall be 160 kbit/s.	
Number of audio channels	Mono and stereo channels shall be supported.	
Error concealment	Shall only rely on information that a packet is lost.	