Technical Specification Group Services and System Aspects **TSGS#21(03)0433**
Meeting #21, Frankfurt, Germany, 22-25 September 2003

Source: TSG-SA WG4 Chairman

Title: TSG-SA WG4 Status Report at TSG-SA#21

Document for: Information

Agenda Item: 7.4.1

Executive Summary

Since TSG-SA#21, TSG-SA WG4 (SA4) has met twice: in July (SA4#27) and in September (SA4#28).

Release 5

Extended Transparent End-to-End PS Streaming Service: The only remaining new Rel-5 specification, "non-critical" TR 26.937 (RTP Usage Model), has been finalised and is brought for approval.

Release 6

Performance characterisation of default codecs for PS conversational multimedia applications: Test plans for the default codecs (AMR and AMR-WB) have been finalised and are brought for information. Subjective tests are to be conducted from September until November with ARCON, France Telecom R&D and NTT-AT acting as subjective testing laboratories. Siemens and France Telecom R&D (acting as host laboratory) will provide the test bed. IPv6 is included in the testing.

PS Streaming Rel-6: Work is ongoing e.g. on bit-rate adaptation, quality metrics and consideration of new codecs. Selection process for audio codecs has been continued (see next section for details). Enhanced video codecs have been discussed and the properties of two codecs (MPEG-4 AVC/ITU-T H.264 and Microsoft WMV9) have been reviewed. A formal video codec selection process is planned and a preliminary version of candidate qualification criteria has been prepared. In order to elaborate the experiments and the effort required for video codec selection (which depend on the number of candidates), all codec proponents must notify the SA4 Secretary of their intention to submit a candidate video codec. This notification must be done by letter by October 3rd by all candidates (including the codecs discussed already earlier in SA4).

Audio codecs (PSS/MMS default audio codecs, extended AMR-WB codec): Test and processing plans have been finalised and are brought for information: "AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan" and "PSS/MMS High-Rate Audio Selection Test and Processing Plan". Funding of the testing among the proponents has been agreed and a document on this is presented for information. The tests (including selection tests and post-selection characterisation tests) are funded by the codec proponents with total cost of 487.5 kEuro. Altogether 8 listening laboratories worldwide will participate in the testing. One critical permanent document "PSS/MMS Audio Codec and Extended AMR-WB Selection Rules" remains still unfinished due to an outstanding issue in the format of the audio codec specifications.

Speech Recognition and Speech Enabled Services: Codec Work to Support Speech Recognition Framework for Automated Voice Services: Recommendation criteria have been finalised and are presented for approval. Updated test and processing plan is brought for information. TSG-SA#20 requested SA4 to assess the candidate codecs' ability to reconstruct speech. SA4 recognizes that both SES codec candidates are capable of reconstructing intelligible speech (on grounds of existing DSR test results from ETSI Aurora and existing AMR/AMR-WB test results in 3GPP). Reconstruction quality of codec candidates will be measured by interested SA4 companies but for informative purposes only.

Media Codecs and Formats for IMS Messaging and Presence: A first skeleton working draft of TS 26.141 "IMS Messaging and Presence; Media formats and codecs" has been prepared. This is based on using the legacy media types (and codecs) from MMS as starting point.

Definition of teleservice using MBMS: Preparation of TS on "MBMS Protocols and Codecs TS" has been started with an initial skeleton draft; the work guided by the ongoing SA1 requirements work. A draft Stage 1 from SA1 was presented for information at SA4#28. A LS was received from SA1 explaining that a full definition of MBMS teleservice is not needed to fulfil the targets of the SA1/SA4 MBMS work. Consequently, SA1 asked SA4 to modify the WID to reflect this (with suggested revisions). SA4 agreed and is bringing an updated WID for approval. In SA4's opinion, a joint meeting between SA1, SA2 and the involved RAN and GERAN WGs might be beneficial to focus and synchronise the overall MBMS work.

WIDs: WID on "Definition of teleservice using MBMS" has been updated (on SA1 request).

Maintenance of releases

Rel-5 CRs to TSs 26.073, 26.132, 26.173, 26.204, 26.234, 26.236, 26.976 and 28.062 are brought for approval.

TSG-SA WG4 Status Report at TSG-SA#21 - Table of Contents

1. General issues

This document presents the status report of TSG-SA WG4 (SA4) at TSG-SA#21. Slides presentation of the report is found in a separate file attached to this status report: “Annex 1 - SA4 Slides Presentation at TSG-SA#21.ppt”.

1.1 Officials

SA4 Vice Chairmen election took place at SA4#27 in July. Two new SA4 Vice Chairmen were elected. Otherwise, there are no changes in the SA4 officials.

The SA4 officials are:

Chairman: Kari Järvinen (Nokia / ETSI)

Vice Chairmen: Catherine Quinquis (Orange, ETSI) and Frédéric Gabin (NEC Technologies, ETSI)

Secretary: Paolo Usai (3GPP Support)

SWG Chairmen:

PSM (Packet Switched Multimedia): Rolf Hakenberg (Panasonic / ETSI)
SQ (Speech Quality): Paolo Usai (ETSI)

Audio codec issues (of WIs PSS Rel-6 and Extended AMR-WB) have been jointly progressed in Audio Codec ad-hoc group. Imre Varga (Siemens / ETSI) has been chairing this ad-hoc group.

1.2 Meetings

Since TSG-SA#20, SA4 has held two plenary meetings SA4#27 (in July) and SA4#28 (in September). In addition, one SQ SWG teleconference on the recommendation criteria for Speech Enabled Services (SES) codecs was held (in mid-June).

Meetings held:

SA4#27	7 – 11 July, 2003	Host: Siemens; Venue: Munich, Germany
SA4#28	1 – 5 September, 2003	Host: Fraunhofer Institute; Venue: Erlangen, Germany

Calendar of future meetings:

SA4 Audio Codec Ad-Hoc (tbc)	7-9 October, 2003	Host: tbd; Venue: tbd
SA4 Video Codec Ad-Hoc	27-29 October, 2003	Host: tbd; Venue: tbd
SA4#29	24 – 28 November, 2003	Host: Nokia; Venue: Tampere, Finland
SA4#30	23 - 27 February, 2004	Host: tbd; Venue: tbd
SA4#31	17 - 21 May, 2004	Host: tbd; Venue: tbd
SA4#32	16 - 20 August, 2004	Host: tbd; Venue: tbd
SA4#33	22 - 26 November, 2004	Host: tbd; Venue: tbd

During SA4#27 and SA4#28, the PSM and SQ SWGs and the audio codec ad-hoc group met. Table 1 gives statistics from these meetings (including also statistics from previous SA4 meetings in 2003 for comparison).

Meeting	Number of (new) input documents	Number of participants	Number of incoming LSs	Number of outgoing LSs/communications
SA4#25	115	55	13	9
SA4#25bis	164	50	14	11
SA4#26	171	55	18	17
SA4#27	**142**	**65**	**19**	**14**
SA4#28	**128**	**55**	**18**	**9**

Table 1: Statistics of recent SA4 meetings

1.3 Input documents from SA4 to TSG-SA#21

Table 2 gives a complete list of SA4 input documents to TSG-SA#21. Some permanent documents[1] on codec selection and testing are brought for information in Tdocs SP-030434 through SP-030439. The recommendation criteria for default codec for SES is brought for approval in Tdoc SP-030440. An updated

[1] Permanent documents are used to guide the detailed work within SA4. These are SA4 documents that are updated during SA4 meetings under version control. They are used, e.g., in new codec development and codec testing to define important aspects such as codec selection test plans, design constraints and performance requirements.

WID on “MBMS user service” and a new TR on “RTP Usage Model” are brought for approval in Tdocs SP-030442 and SP-030443. Several Rel-5 CRs are presented for approval in Tdocs SP-030444 through SP-030451. In addition, a LS on SES codecs ability to reconstruct speech is presented in Tdoc SP-030369.

Tdoc	Title	Source	Agenda item	Document for
SP-030369	LS (from SA WG4) on "Assessment of the SES codecs ability to reconstruct speech"	SA WG4	7.4.1	Information
SP-030433	TSG S4 Status Report at TSG-SA#21	SA WG4 Chairman	7.4.1	Information
SP-030434	Test and processing plan for default codec evaluation for speech enabled services (SES)	SA WG4	7.4.1	Information
SP-030435	Test Plan for the AMR Narrow-Band Packet Switched Conversation test	SA WG4	7.4.1	Information
SP-030436	Test Plan for the AMR Wide-Band Packet Switched Conversation test	SA WG4	7.4.1	Information
SP-030437	AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan	SA WG4	7.4.1	Information
SP-030438	PSS/MMS High-Rate Audio Selection Test and Processing Plan	SA WG4	7.4.1	Information
SP-030439	Funding of Audio Codec Testing	SA WG4	7.4.1	Information
SP-030440	Recommendation Criteria for Default Codec for Speech Enabled Services (SES)	SA WG4	7.4.3	Approval
SP-030442	Updated Work Item Description on Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS) (Release 6)	SA WG4	7.4.3	Approval
SP-030443	TR 26.937 on "RTP usage model" v. 2.0.0 (Release 5)	SA WG4	7.4.3	Approval
SP-030444	CR to TS 26.073 - Correction of the MMS_IO flag (Release 5)	SA WG4	7.4.3	Approval
SP-030445	CR to TS 26.132 - Loudness rating measurements at lower bit rates (Release 5)	SA WG4	7.4.3	Approval
SP-030446	CR to TS 26.173 - Possible decoder LPC coefficients overflow (Rel 5)	SA WG4	7.4.3	Approval
SP-030447	CR to TS 26.204 - Possible decoder LPC coefficients overflow (Release 5)	SA WG4	7.4.3	Approval
SP-030448	CRs to TS 26.234 - Corrections (Release 5)	SA WG4	7.4.3	Approval
SP-030449	CRs to TS 26.236 - Corrections (Release 5)	SA WG4	7.4.3	Approval
SP-030450	CR to TS 26.976 - Reference to incorrect test results (Release 5)	SA WG4	7.4.3	Approval
SP-030451	CR to TS 28.062 - Removal of Pre-Handover Notification for UMTS (Release 5)	SA WG4	7.4.3	Approval

Table 2: List of input documents to TSG-SA#21 from SA4

1.4 List of Annexes

This report contains the following Annexes (all in separate files):

1) Annex 1 - SA4 Slides Presentation at TSG-SA#21
2) Annex 2 - S4-030659 “Advanced video codec selection: Candidate requirements (draft)”
3) Annex 3 - S4-030703 “Draft PSS/MMS Audio Codec and Extended AMR-WB, Selection Rules (Version 0.4)”

2. Remaining Release 5 work

TR 26.937 “RTP usage model” is the only remaining new Rel-5 specification still to be produced in SA4. This non-critical TR brings additional information to characterise the PS Streaming Service (e.g., give statistics of traffic characteristics such as packet sizes and bit-rates) and also gives useful information on issues that service providers and manufacturers should be aware of (e.g., implications of chosen RTP packet sizes and impact of different rate control strategies for video streaming).

Version 1.2.0 of the TR was presented for information at TSG-SA#18 (in Tdoc SP-020683). Since then, updated versions have been prepared and feedback received from the relevant WGs (e.g., SA1, SA2, RAN2 and GERAN). The TR has now been finalised and is brought for approval (in Tdoc SP-030443).

3. Release 6 Work Items

3.1 Performance Characterisation of Default Codecs for PS Conversational Multimedia Applications

Test Plans covering the AMR narrowband and the AMR wideband (AMR-WB) codecs have been finalised and are brought for information in Tdocs SP-030435 and SP-030436. Conversational (bi-directional) testing will be used to realistically capture the quality (and degradations) experienced during conversations via the PS domain. (Uni-directional listening-only tests have been used in previous codec characterisations described in the existing codec performance characterisation TRs of 26-series.) The tests consist of 24 test conditions both for the AMR and the AMR-WB codec. These contain error conditions with IP packet loss of 0% and 3% and under BER ratios 10^{-2}, 10^{-3} and 5 10^{-4}. Robust Header Compression (RoCH), an optional UMTS functionality, is included for some test cases for AMR-WB tests. Three types of background noise are used: car, street and cafeteria. The codec modes 6.7 and 12.2 kbit/s are included in the tests for AMR, and the modes 12.65 and 15.85 kbit/s for AMR-WB.

The real-time test bed (UMTS simulator) is provided by Siemens and France Telecom R&D. It consists of altogether 5 PCs: two PCs simulating the two terminals, two PCs simulating the two air-interfaces, and one PC performing the network simulation. IPv6 is used in the characterisation. IPv6 is simulated fully over the radio interface to ensure that the impact of bit-errors is properly simulated for the IPv6 protocol. For the core network simulator, IPv4 is used instead of IPv6. However, as the core network simulator handles only packets (and not bits or bytes in packets), the use of IPv4 or IPv6 is not relevant in this part of the simulator. Using IPv4 or IPv6 has no impact for characterisation purposes. The reason for using partly IPv4 is that the available core network simulator supports only IPv4. The only impact is a minor difference in the bit-rate transmitted in the core network resulting into negligible difference in the end-to-end delay (in the order of ~16 µs).

The subjective tests are scheduled in the time frame from mid-September until late November 2003. ARCON, France Telecom R&D, and NTT-AT will act as subjective testing labs. Contracts for the involved laboratories have been finalised. Budget "up to 194 kEuro" has been allocated earlier for this exercise (160 kEuro allocated by 3GPP PCG and the contingency of “up to 34 kEuro” left from the AMR-WB Characterisation Phase). A compensation of 40 kEuro for each individual laboratory has been agreed, including testing and preparation of reports. The contingency of maximum 34 kEuro will be utilized for global analysis of results, with the remaining part to be used for further characterisation testing (details to be decided in SA4). A written confirmation by letter for authorizing the use of the AMR-WB contingency is still awaited from three companies.

Table 3 lists the output specification for this WI (one TR). A draft TR could be expected for information at TSG-SA#22. Completion of the TR is targeted at TSG-SA#23.

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TR 26.9yz	Performance characterization of default codecs for PS conversational multimedia applications	SA4	-	TR preparation is pending on the testing.	TSG-SA#23 (March 2004)

Table 3: Status list of output TSs/TRs/CRs for Performance characterisation of default codecs for PS conversational multimedia applications

3.2 Packet Switched Streaming Rel-6 (excluding consideration of new audio codecs)

Rel-6 streaming has been discussed further with most debate on bit-rate adaptation (used to avoid quality degradations under critical conditions, e.g., due to network bandwidth variability and service interruptions such as handovers), on quality metrics (for servers to receive from the handset to provide the service providers means to evaluate the end user experience), and on consideration of new codecs (for audio and also for video).

On bit-rate adaptation, SA4#28 agreed that no particular adaptation algorithm will be mandated (for PSS clients). On streaming service quality metrics, some updates were agreed e.g. for the RTSP protocol definition to send the quality metrics. SA4 internal “PSS Quality metrics” permanent document was updated accordingly (into v.0.06). A basic set of quality metrics have been agreed in SA4 earlier. GZIP technology has been chosen as the working assumption for the compression of SVG (Scalable Vector Graphics) content for Rel-6. Compression of SVG content allows more efficient usage of network bandwidth. Updated working drafts were produced for the streaming TSs 26.234 (v.0.3.0), 26.244 (v.0.2.5), 26.245 (v.0.1.6) and 26.246 (v.0.0.3).

Selection process for PSS/MMS audio codecs has been continued (see section 3.3).

The consideration of enhanced video codecs for Rel-6 was raised at SA4#27. The characteristics of two codecs (MPEG-4 AVC/ITU-T H.264 and Microsoft WMV9) have since then been initially reviewed in SA4, and a formal video codec selection process is planned. A preliminary version of candidate qualification criteria including preliminary time plan of the selection process has been prepared. This is provided for information in Annex 2 of this status report (in a separate document). In order to elaborate the experiments and the effort required for the codec selection, all proponents (including the initial proposals made earlier at SA4 meetings) must notify via a letter the SA4 Secretary of their intention to submit a candidate video codec. This notification must be given by October 3rd 2003 (24:00 CET) in order to be considered in the selection process. For planning purposes it is necessary to know the maximum number of codecs at an early phase of the selection process.

To progress the video codec selection process in a timely manner, a SA4 Video Codec Ad-Hoc meeting is scheduled on 27-29 October (with decision power given by SA4 to finalise and approve the video codec candidate qualification criteria). Candidate qualification material (as defined in the finalised qualification criteria document) must then be submitted by each candidate for SA4#29 meeting in November. The video codec selection is planned to take place at SA4#30 in February 2004.

Dialogue has been continued with OMA DLDRM working group regarding interworking issues between OMA DRM and 3GPP PSS Rel-6 service, e.g., verifying suitability of OMA DRM v1.0 for progressive download in PSS. SA3 has also been involved in the LS exchange between SA4 and OMA DLDRM, and SA4 understanding is that encryption will be done by SA3.

Several LSs have been sent to other groups regarding Rel-6 streaming issues. These include a LS to W3C requesting information on future extensions of SVG currently under development in W3C. MPEG and ITU-T SG16 have been communicated SA4 request to include higher maximum resolutions and bit-rates in H.264/AVC video codec making it better suitable for 3GPP use (e.g., for streaming of action rich content). IMTC has yet again identified in incoming LSs some valuable interoperability issues that have then been resolved in SA4 (by CRs). IMTC has been thanked and informed on the resulting SA4 actions.

Table 4 lists the intended PSS Rel-6 output specifications (including Stage 1 and Stage 2 since also these are covered in the SA4 lead PSS Rel-6 Work Item).

Deliverable	Title	Prime resp. WG	2nd resp. WG?	Comment/Status	TSG-SA approval target
CRs to TS 26.233	Transparent end-to-end PSS; General description	SA4	SA2	To be updated based on the content of PSS Rel-6.	TSG-SA#23 (March 2004)
CRs to TS 26.234	Transparent end-to-end PSS; Protocol and codecs	SA4	SA2	Working draft (v.0.3.0) exists in SA4	TSG-SA#23 (March 2004)
TS 26.244	Transparent end-to-end PSS; File Format	SA4	SA2	Working draft (v.0.2.5) exists in SA4	TSG-SA#23 (March 2004)
TS 26.245	Transparent end-to-end PSS; Timed Text Format	SA4	SA2	Working draft (v.0.1.6) exists in SA4	TSG-SA#23 (March 2004)
TS 26.246	Transparent end-to-end PSS; SMIL Language Profile	SA4	SA2	Working draft (v.0.0.3) exists in SA4	TSG-SA#23 (March 2004)
CRs to TS 22.233	Stage 1	SA1		Under SA1 responsibility.	
Possible new TS	Stage2 (non-transparent aspects)	SA2		To be produced by SA2, if needed.	

Table 4: Status list of output TSs/TRs/CRs for Packet Switched Streaming Rel-6

TS 26.140 "MMS Media formats and codecs" working draft (v.0.1.1) has been updated based on the developments in PSS Rel-6 to the extent this MMS specification is impacted (e.g., split of TS 26.234 in Rel-6 into four separate TSs, GZIP technology chosen as the working assumption for the compression of SVG).

3.3 Audio Codecs (PSS/MMS default audio codecs, and extended AMR-WB codec)

Note: This section reports jointly the related audio codec work done within Work Item "PSS Rel-6" (consideration of new audio codecs for PSS and MMS) and within Work Item "Extended AMR-WB codec" (development of AMR-WB+ codec, a candidate codec for PSS/MMS audio).[2]

[2] The work in both WIs is very related as the AMR-WB+ codec is considered as one candidate for PSS/MMS default audio codec, and the testing of codec candidates for both WIs will be carried out as combined testing. (The detailed audio codec work for both WIs has been progressed jointly by SA4 audio codec ad-hoc group.)

3.3.1 Test and Processing Plans and Agreement on funding

As explained earlier to TSG-SA, the low bit-rate audio range (12 kbit/s to < 32 kbit/s) and the high bit-rate audio range are tested in separate experiments. In order to allow some flexibility in defining the exact breaking point between the low and high bit-rate range, an area of overlap of the two bit-rate ranges is introduced in the testing. This is achieved by considering the results for 24 kbit/s for both the low and the high bit-rate range. The low bit-rate range is intended for speech, music and mixed content, while the high bit-rate range is intended specifically for music. This is seen in the selection of test material for the tests.

Test and processing plans have now been finalised and are brought for information to TSG-SA#21. These are “AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan” (in Tdoc SP-030437), and “PSS/MMS High-Rate Audio Selection Test and Processing Plan” (in Tdoc SP-030438).

The AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test is divided into two sets of experiments: Block A (Intrinsic quality comparison of candidate codecs) and Block B (Quality comparison under stressed operating conditions). The detailed content of these is as listed below:

- Experimental block A:
    - 14 kbps, mono, use case A (PSS)
    - 18 kbps, stereo, use case A (PSS)
    - 24 kbps, mono, use case A (PSS)
    - 24 kbps, stereo, use case A (PSS)
- Experimental block B:
    - 14 kbps, mono, use case B (MMS), 16 kHz input and output sampling rate.
    - 18 kbps, stereo, use case B (MMS)
    - 14 kbps, mono, use case A (PSS), 3% FER
    - 24 kbps, stereo, use case A (PSS), 3% FER

The PSS/MMS High-Rate Audio Selection Tests consist of three experiments with the following operational conditions:

- Experiment 1: 32 kbps, stereo, use case A (PSS) and use case B (MMS)
- Experiment 1: 48 kbps, stereo, use case A (PSS) and use case B (MMS)
- Experiment 1: 32 kbps, stereo, use case A (PSS) with 1% and 3% random frame loss

Altogether eight listening laboratories will participate in the AMR-WB+ and PSS/MMS Low-Rate testing and six for the PSS/MMS High-Rate testing. The following listening laboratories will participate: T-Systems, NTT-AT, France Telecom R&D, Dynastat, Nokia, Ericsson, Coding Technologies, and Fraunhofer Institute. Each test condition will be tested twice by two different laboratories. T-Systems and Audio Research Labs will act as host laboratories (to process the test material). Each processing is done twice in the two different laboratories to cross check the processing.

Audio material used for AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test is classified according to the following four content types: Speech, Speech over Music, Speech between Music, and Music. Audio material used for PSS/MMS High-Rate Audio Selection Test consists of the following categories: pop (with and/or without vocals), classic (with and/or without vocals), single instruments, a capella vocals (solo and/or choir), mixed speech and music, and speech with and/or without background noise. The test material will be prepared by sending first a call out for test material according to a pre-defined number of generic audio signal categories. Then, an independent selection entity will choose material representative for assumed typical applications to be used in the selection tests.

The listening tests are funded by the codec proponents with total funding of 400 kEuro for the AMR-WB+ and PSS/MMS Low-Rate tests and 87.5 kEuro for the PSS/MMS High-Rate tests. The funding is detailed in “Funding of Audio Codec Testing” document brought for information in Tdoc SP-030439.

Table 5 shows the codec candidates (and the reference codecs) to be included in the audio codec tests.

#	Candidate Codec	AMR-WB+ candidate	PSS/MMS low-rate audio candidate	Supporting/providing Organization(s)
1	MPEG4 HE-AAC codec ("aacPlus")	No	Yes	Coding Technologies, NEC, Panasonic
2	AMR-WB+ candidate codec	Yes	Yes	Ericsson, Nokia, VoiceAge
3	CT codec ("Enhanced aacPlus")	No	Yes	Coding Technologies
#	**Reference codec**	-	-	**Providing Organization(s)**
5	AAC	-	-	Fraunhofer Institute
6	AMR-WB	-	-	3GPP

Table 5a: Candidate and reference codecs for AMR-WB+ and PSS/MMS Low-Rate testing

#	Candidate Codec	Supporting/providing organization(s)
1	MPEG4 HE-AAC codec ("aacPlus")	Coding Technologies, NEC, Panasonic
2	CT codec ("Enhanced aacPlus")	Coding Technologies
#	**Reference codec**	**Providing Organization(s)**
3	AAC	Fraunhofer Institute
4	RealAudio (for informative purposes)	RealNetworks

Table 5b: Candidate and reference codecs for PSS/MMS High-Rate testing

Operational modes/conditions not covered by selection tests, for which performance requirements are defined, will be tested during characterization of the selected codec.

3.3.2 Selection Rules: outstanding issue on Specification Format

The remaining permanent document on “PSS/MMS Audio Codec and Extended AMR-WB Selection Rules” has been progressed. A rather mature draft version was prepared during SA4#28 based on the methodology used at earlier codec selections in 3GPP (e.g., for AMR-WB codec). Two eliminating rules would be used first to exclude all candidates failing to demonstrate compliance with the PSS/MMS audio codec design constraints or presenting test results too far below the required performance level. A set of primary Figures of Merit is given according to which the performances of the remaining candidates would then be compared as part of the selection process. However, the Selection Rules document could not be finalised and agreed in SA4 in time for TSG-SA#21 due to an outstanding issue in the format of the audio codec specification.

There has been a debate in SA4 on how detailed description of the “winning codec” will be included in the specifications, and specifically on if the tested ANSI-C code of the PSS and MMS encoders will be included in the codec specifications or if minimum quality level could be guaranteed by defining normative minimum quality level specification (with the winner also obliged to make a reference source code, as tested, available outside specifications upon request under fair, reasonable and non-discriminatory terms and conditions).The following alternatives how to proceed were identified in extensive discussions during SA4#28 in the Audio Codec ad-hoc group (from the audio codec ad-hoc report):

- PSS and MMS decoders:
    - Specify decoder in each case by ANSI-C source code as integral and normative part of the specification and maintain it by usual 3GPP change control. Attach test vectors to be able to test compliance of implementations.
- PSS encoder:
    - Method A: Specify PSS encoder in each case by ANSI-C source code as integral part of the specification and maintain it by usual 3GPP change control. Attach test vectors to be able to test compliance of implementations.
    - Method B: The normative specification of the encoder is a minimum quality level (description based on selection phase results). The meaning is that all Rel6 audio PSS compatible implementations shall not perform worse than this minimum quality level. No source code is attached. The winner is obliged to make a reference source code as tested, available upon request under fair, reasonable and non-discriminatory terms and conditions. Informative executable is attached. 3GPP SA4 is authorized to endorse implementations of the PSS encoder for compliance with the specification, according to a pre-defined test plan.

- Method C: require proponents to declare their policy as part of selection deliverables whether they follow PSS encoder Method A or Method B. Give preference in selection rules to proponents supporting Method A.
- Method D: require proponents to declare their policy as part of selection deliverables whether they follow PSS encoder Method A or Method B. Take the choice into account in selection rules.

- MSS encoder: (the same alternatives identified as for PSS encoder above)

The above approach for PSS and MMS decoders was agreed, but no consensus could be reached in the audio codec ad-hoc group between the above methods for the PSS and MMS encoders. When the issue was further debated in the SA4#28 Plenary session, the following wording (for the Selection Rules) was prepared as potential compromise. It states that the audio codec specifications will contain detailed technical description of the encoder and decoder and the ANSI-C source code of the tested encoder and decoder, but that the access to the source code shall be restricted to eligible entities:

> "The format of the specification is as described below:
>
> - Detailed technical description of the encoder and decoder
> - ANSI-C source code of the tested encoder and decoder
>
> Source code for both encoder and decoder (in the form as tested) shall be under the copyright and control of 3GPP (or one of its organizational partners) and shall be available from the controlling entity (3GPP or one of its organizational partners) to interested companies having a bona fide interest in receiving such source code.
>
> Diligent efforts will be made to establish an infrastructure of confidential disclosure of the software, in order to ensure that valuable implementation details and trade secrets are not used by non-eligible entities, not used for products not compliant with the 3GPP specification and not used for non-3GPP applications. Access to the software may be subject to a reasonable fee paid to the controlling entity and will be granted solely for applications within the scope of the respective 3GPP specification. (Wording to be reviewed by 3GPP legal adviser.)"

Selection Rules document with the above wording was prepared as outcome of SA4#28 (1-5 September 2003) in Tdoc S4-030703 "Draft PSS/MMS Audio Codec and Extended AMR-WB, Selection Rules Version 0.4". This document is provided for information in Annex 3 of this status report (in a separate document). Orange requested at SA4#28 to leave time to consider the proposed wording. Also, the wording was felt needing review by 3GPP legal adviser (as indicated in the text itself) since part of the text concerns non-technical matters NOT under SA4 responsibility. This Selection Rules document was therefore put for e-mail approval until September 12th.

The subsequent e-mail approval on SA4 reflector resulted in two objections (from France Telecom / Orange and Nokia), both on the proposed limited access to the source code. France Telecom / Orange pointed out two issues: 1) A definition of "non-eligible entity" (or of "eligible entity") is necessary; and 2) What is the purpose of the fee? Some discussion on these issues followed on the SA4 reflector, but the France Telecom / Orange objection was sustained, claiming an apparent contradiction between the fact that ETSI cannot discriminate (to whom to deliver the source code) and that there could be non-eligible entities. The Legal Adviser at ETSI, Mr. Stephane Tronchon, stated (consulted by SA4 Secretary) that the text of SA4 document is basically fine 'as is', but that it needs to be interpreted taking into account further explanations/clarifications and existing similar cases where ETSI is acting as the Custodian of source codes. As a result of the discussion on SA4 reflector and the given interpretations of eligible and non-eligible entities, also Nokia expressed that they do not accept the text. The Selection Rules were thus not approved by SA4 and therefore could not be finalised by TSG-SA#21 as planned.

SA4 has scheduled an audio codec ad-hoc meeting to take place in early October (October 7-9, tbc) in case Selection Rules were not approved by correspondence (which then happened). SA4 has given decision power to the ad-hoc meeting to finalise and approve the Selection Rules in order to make progress. If agreement on the specification format will not be reached at/by the audio codec ad-hoc meeting, it could at the worst jeopardise Rel-6 schedule for the PSS/MMS audio codec work. The subjective testing for the PSS/MMS audio codec selection has been scheduled for December-January timeframe with first steps for the testing to be taken already during October (submission of candidate codecs and preparation of the test material and error patterns for the testing).

Advice is sought from TSG-SA on the above matter remaining unsolved in SA4.

Table 6 lists the intended output specifications and their work status.

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
CRs to 26-series AMR-WB TSs/TRs	(Relevant AMR-WB specifications of 26 –series)	SA4	-	Drafts to be prepared by candidates for the codec selection meeting (SA4#30).	TSG-SA#23 (March 2004)
New audio codec TS(s)		SA4	-	Drafts to be prepared by the candidates for the codec selection meeting (SA4#30)..	TSG-SA#23 (March 2004)
CRs to TS 26.234	Transparent end-to-end PSS; Protocols and codecs	SA4	SA2	Default codec definition for the audio media type needs to be updated based on the selection of new audio codec(s).	TSG-SA#23 (March 2004)

Table 6: List of output TSs/TRs/CRs for audio codecs (for WIs "PSS Rel-6" and "Extended AMR-WB codec")

3.4 Speech Recognition and Speech Enabled Services: Codec Work to Support Speech Recognition Framework for Automated Voice Services

The remaining permanent document on "Recommendation Criteria" has been completed and is presented for approval in Tdoc SP-030440. The criteria is intended to be used for making recommendation between the two SES default codec candidates: 1) DSR AFE (ETSI DSR standard ES 202 050) and its extension and 2) the AMR and AMR-WB codecs. The criteria consists of first excluding candidates not compliant with all design constraints. Then, the candidates will be compared against each other based on relative reduction in average word error rate. The AMR/AMR-WB codec mode used in these comparisons is the AMR 4.75, AMR 12.2, and AMR-WB 12.65. In case the relative reduction for the DSR AFE codec and its extension compared to the AMR 4.75/ AMR 12.2/ AMR-WB codec is more than 35% / 30% / 25%, respectively, then the DSR AFE codec and its extension will be recommended. In case the reduction is less than 20% /20% / 15% then the AMR/AMR-WB codec will be recommended. With results in between these limits (i.e. 20% and 35%, 20% and 30%, and 15% and 25%), the performance results will be further considered by SA4 (and if there is no consensus the results will be passed to TSG-SA for decision on what recommendation to make). This procedure for analysing the relative reduction in average word error rate is carried out for low-rate data comparison at 8 kHz sampling rate and for high data-rate comparison at 8 kHz and at 16 kHz.

"Test and Processing Plan" is presented for information in Tdoc SP-030434. The recognition test experiments used in the codec evaluation cover a range of tasks: connected digit recognition task, sub-word trained model recognition task and tone confusability task. Testing is done in error-free channel as well as under packet loss situations. The channel error experiments cover average channel BLock Error Rates (BLER) of 1% and 3%. In addition, a BLER of 10% is tested for informative purposes. The testing covers Error Patterns (EP) for UTRAN and EGPRS/GPRS channels. AMR modes of 4.75 and 12.2 are included in the tests for 8 kHz sampling rate test case, and AMR-WB modes of 12.65 and 23.85 for 16 kHz sampling rate case. Two Automatic Speech Recognition (ASR) vendors, IBM and SpeechWorks (now ScanSoft), will carry out the testing (on voluntary basis). Preparation for the tests is currently ongoing. Both vendors run tests for both candidates. The vendors have a free choice over the recogniser back-end configuration. The results of evaluations will be reported to SA4#29 for making recommendation based on the agreed recommendation criteria. The codec selection would then be brought for approval at TSG-SA#22 (December).

TSG-SA#20 requested SA4 to assess the codecs' ability to reconstruct speech. As a response, SA4 recognizes that both SES codec candidates are capable of reconstructing intelligible speech (on grounds of existing DSR test results from ETSI Aurora and existing AMR/AMR-WB test results in 3GPP). Reconstruction quality of codec candidates will be measured by interested SA4 companies but for informative purposes only. A LS on this is presented in Tdoc SP-030369.

At SA4#28, a LS was received from SA2 on "Usage of Speech Enabled services in CS Domain" to which SA4 responded that SA4 have no data available currently to quantify the potential improvement by the mentioned methods for CS domain (indication of codec type for the speech recogniser, introduction of specific speech recognition mode in the terminal), and that this would require further investigation which is not currently on-going in SA4. It was also explained that SA4 is currently working on SES for PS domain, which is expected to provide some generic indications of the factors that determine speech recognition performance. SA4 will keep SA2 informed on the results of this work if relevant for the speech recognition performance in CS domain.

Table 7 lists the intended output specifications and their work status.

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
CRs to TS 26.235	PS Conversational Multimedia Applications; Default Codecs	SA4	SA2, T2	To be prepared based on the codec selection.	At the earliest at TSG-SA#22 (Dec 2003)
CRs to TS 26.236	PS Conversational Multimedia Applications; Transport Protocols	SA4	SA2, T2	To be prepared based on the codec selection	At the earliest at TSG-SA#22 (Dec 2003)
Possible new TSs	Codec specification	SA4		To be prepared, if needed.	At the earliest at TSG-SA#22 (Dec 2003)

Table 7: Status list of output TSs/TRs/CRs for Codec Work to Support Speech Recognition Framework for Automated Voice Services

3.5 Media Codecs and Formats for IMS Messaging and Presence

A first “skeleton” working draft of TS 26.141 “IMS Messaging and Presence; Media formats and codecs” has been prepared. This is based on using the legacy media types (and codecs) from MMS as starting point for IMS Messaging and Presence. The discussion of new audio and video codecs for PSS/MMS in Rel-6 ongoing in SA4 will be taken into account in the work. The emerging Common Presence and Instant Messaging Message format CPIM (details being speficied by CN1 in TS 24.841) and the technical requirements following from this are to be considered within the SA4 work.

Table 8 lists the status of intended output specification.

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TS 26.141	IMS Messaging and Presence; Media formats and codecs	SA4	SA2, CN1	First skeleton working draft prepared.	TSG-SA#23 (March 2004)

Table 8: Status list of output TSs/TRs/CRs for Media Codecs and Formats for IMS Messaging and Presence

3.6 Definition of teleservice using MBMS

Preparation of the SA4 TS on “MBMS Protocols and Codecs TS” has been started in SA4 with an initial “skeleton” draft; the work guided by the ongoing SA1 requirements work. A draft SA1 TS on Stage 1 on “MBMS User Services” from SA1 was presented for information at SA4#28. SA1 also explained to SA4 that a full definition of MBMS teleservice is unnecessary to fulfil the targets of the work, and consequently, has asked SA4 to modify the WID to reflect this (with suggested revisions). SA4 agreed on the revisions and is bringing a revised WID for approval.

As a response to RAN2 and RAN3 for LS on scalable codecs on MBMS, SA4 has responded that SA4 sees scalable codecs not being within the scope of Rel-6 MBMS, and that SA4 will eventually consider them in future MBMS releases. SA4 work is currently focusing on studying codecs and protocols for the basic MBMS service, and these need to be settled before any more advanced feature is considered.

In SA4’s opinion, a joint meeting between SA1, SA2 and the involved RAN and GERAN WGs might be beneficial to focus and synchronise the overall MBMS work. (The relevant WG Chairmen will be contacted off-line to find out if there is any interest or possibilities for such a joint meeting.)

Table 9 lists the status of the intended output specification.

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TS 26.x.y.z	MBMS Protocols and Codecs	SA4	SA2, SA3	First skeleton working draft prepared.	TSG-SA#23 (March 2004)

Table 9: Status list of output TSs/TRs/CRs for Definition of teleservice using MBMS

4. Work Item Descriptions

4.1 Revised Rel-6 WID on "Definition of teleservice using MBMS"

The joint SA4/SA1 WID on "Definition of teleservice using MBMS" approved at TSG-SA#20 has been updated into "Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS)" (on request by SA1). See details in Section 3.6.

The revised WID is brought for approval in Tdoc SP-030442.

5. Communication with other WGs/TSGs/groups

Table 10 gives a complete list of LSs sent out (to other WGs/TSGs and 3GPP external groups) since TSG-SA#20. (The communication to ITU-T has been delivered through company contributions in ITU-T.)

Tdoc no.	Title	Intended for	Copy to
TD S4-030512	(Reply) LS on clarification on minimum set of TFCs for TFC selection	RAN2	RAN4
TD S4-030549	Liaison Statement on Discard Timer	RAN3	SA2, RAN2, GERAN2
TD S4-030550	Liaison response on Meta-Data in ISO Media Files, Streaming Text, Advanced Text and Graphics Amendment	ISO/IEC SC29/WG11	
TD S4-030510	LS on DRM for Progressive Download	OMA MAG/DL+DRM	SA3
TD S4-030518	Reply to "Liaison regarding RTP timestamps"	IMTC PSS AG	
TD S4-030509	Communication / LS on the specified levels in ITU-T Recommendation H.264 | ISO/IEC International Standard 14496-10 (MPEG-4 AVC)	ITU-T SG16 WP3 Question 6 and Joint Video Team; ISO/IEC JTC1/SC29/WG11 (MPEG)	
TD S4-030529	Liaison Statement on specified levels in MPEG-4 Visual Simple Profile	ISO/IEC JTC1/SC29/WG11 (MPEG)	
TD S4-030532	LS on compression of SVG content and progressive downloading	W3C SVG group	
TD S4-030547	LS on SA4 assessment of the SES Codecs ability to reconstruct speech	TSG SA	
TD S4-030558	Reply to LS on Core Network Provision of separate flows for P2P and P2M radio Transmission	CN1, CN4, RAN1, RAN2, GERAN1, GERAN2, SA1, SA2	
TD S4-030573	Reply to LS on future codecs with same SDU format as existing codecs	TSG CN, CN4	
TD S4-030557	Communication to ITU-T on the specified levels in Annex X of ITU-T Recommendation H.263	ITU-T SG16 WP3 Question 6 (Video Coding Experts Group)	
TD S4-030552	LS on DRM issues	OMA MAG/DL+DRM	
TD S4-030576	LS on DRM issues	ISMA	
TD S4-030660	LS on cipher suite for DRM-protected streamed media for PSS	SA3	
TD S4-030647	Liaison Response to OMA on DRM issues	OMA	SA3
TD S4-030683	Communication to ITU-T SG9 on Timed Text	ITU-T SG9	
TD S4-030684	Liaison to 3GPP2 on Timed Text	3GPP2	
TD S4-030670	LS on "Update of WID on MBMS"	SA1	SA2, SA3, SA5, RAN2, RAN3, GERAN1, GERAN2, CN1
TD S4-030685	Response to Liaison Statement on "Reliable transport" for PSS	SA1	
TD S4-030686	Reply to LS on "Usage of RTCP & SDP in MBMS"	RAN2	SA2, RAN3, GERAN1, GERAN2
TD S4-030679	Reply to "Usage of Speech Enabled Services in CS Domain"	SA2	
TD S4-030687	LS on scalable codecs for MBMS	RAN2	RAN3, SA2

Table 10: SA4 LSs sent out since TSG-SA#20

6. Maintenance of Releases

CRs to TSs 26.073, 26.132, 26.173, 26.204, 26.234, 26.236, 26.976 and 28.062 are brought for approval. These are contained in Tdocs SP-030444 until SP-030451 and are summarised in the tables below with some comments. All are for Release 5.

Tdoc SP-030444: 26.073 “ANSI-C code for the Adaptive Multi Rate (AMR) speech codec”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.073	018		Rel-5	Correction of the MMS_IO flag	F	5.1.0	S4	TSG-SA WG4#27	S4-030485

- The code of the AMR codec does not compile when the MMS option is enabled. Therefore, the developers can not test the code with the AMR MIME file storage format unless the bug is corrected.

Tdoc SP-030445: 26.132 “Speech and video telephony terminal acoustic test specification”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.132	026		Rel-5	Loudness rating measurements at lower bit rates	F	5.3.0	S4	TSG-SA WG4#28	S4-030619

- A note is added to explain that the use of multisine signal is not recommended for measurements of loudness ratings at lower AMR bit-rates than 12.2 kbit/s. Although only 12.2 kbit/s mode is defined for use in testing, some manufacturers have done additional testing with the lower bit-rate modes and come up with peculiar results. (For multisine signal, the test results depend very much on the selected AMR bit-rate.)

Tdoc SP-030446: 26.173 “ANSI-C code for the AMR-WB codec”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.173	019		Rel-5	Possible decoder LPC coefficients overflow	F	5.7.1	S4	TSG-SA WG4#28	S4-030634

- AMR-WB decoder can produce unstable output during DTX-operation (source controlled operation). Conversion from ISP to LPC coefficients is changed so that LPC coefficients cannot overflow in decoder comfort noise generation. Otherwise, the decoder synthesis filter may become unstable causing uncontrolled output when DTX is used.

Tdoc SP-030447: 26.204 “ANSI-C code for the Floating-point AMR-WB codec”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.204	008		Rel-5	Possible decoder LPC coefficients overflow	F	5.1.0	S4	TSG-SA WG4#28	S4-030635

- (same reason as above)

Tdoc SP-030448: 26.234 “Transparent end-to-end PSS; Protocols and codecs“

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.234	061	1	Rel-5	Clarification on session bandwidth for RS and RR RTCP modifiers	F	5.5.0	S4	TSG-SA WG4#27	S4-030556
26.234	062	1	Rel-5	Correction of ambiguous range headers in SDP	F	5.5.0	S4	TSG-SA WG4#27	S4-030511
26.234	063	1	Rel-5	Timed-Text layout example	F	5.5.0	S4	TSG-SA WG4#27	S4-030555
26.234	064		Rel-5	Correction of ambiguity in RTP timestamps handling after PAUSE/PLAY RTSP requests	F	5.5.0	S4	TSG-SA WG4#27	S4-030517
26.234	065		Rel-5	Correction of obsolete RTP references	F	5.5.0	S4	TSG-SA WG4#28	S4-030607
26.234	066	1	Rel-5	Correction of wrong reference	F	5.5.0	S4	TSG-SA WG4#28	S4-030648
26.234	067		Rel-5	Missing signaling of live content	F	5.5.0	S4	TSG-SA WG4#28	S4-030654

- These contain the following corrections:
  - Session bandwidth for RS and RR RTCP modifiers are corrected to avoid misinterpretation. Otherwise, the rules are ambiguous and interoperability problems will occur.
  - The overview table of SDP fields indicates that the a=range field is required on both session and media levels, which is inconsistent with the usage of the field explained elsewhere. A note is added explaining the correct usage. Otherwise, media streams of different lengths would be ambiguous and were interpreted differently by different clients thus impacting interoperability.
  - The Timed-Text layout example given is wrong. The character order is corrected. Otherwise, implementations may follow the example resulting in incomprehensible text. (Even though it is an example, it is an important part of the specification. The text is hard (if not impossible) to understand without a correct example.)
  - Interoperability tests have showed that the current specification is unclear about how a PSS server has to timestamp RTP packets after a PLAY request. Text is added clarifying what must be done by the server with RTP timestamps. Otherwise, different interpretations would lead to non-interoperable solutions and prevent the playback control to work properly.
  - Obsolete RTP references have been corrected Otherwise, the mandatory SDP bandwidth modifiers RR and RS cannot be correctly implemented and implementers will encounter problems.
  - The IETF RFC number for Session Description Protocol (SDP) bandwidth modifiers for RTCP bandwidth

is wrong and is now corrected. Otherwise, the reference points to the wrong IETF document and this could result in problems for implementations.

- There are two types of media a PSS server can offer: on-demand and live. In the current PSS specification, there is no explanation on how the server can signal live content and therefore a client will not be able to act accordingly. Explanation on how live content can be signalled is included. Otherwise, a client will not be aware that the offered content is live, which may lead to interoperability problems and failure of service.

Tdoc SP-030449: 26.236 “PS conversational multimedia applications; Transport protocols“

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	006		Rel-5	Correction of obsolete RTP references	F	5.3.0	S4	TSG-SA WG4#28	S4-030608
26.236	007	1	Rel-5	Correction of wrong reference	F	5.3.0	S4	TSG-SA WG4#28	S4-030649

- Obsolete RTP references have been corrected Otherwise, the mandatory SDP bandwidth modifiers RR and RS cannot be correctly implemented and implementers will have problems resolving ambiguities.
- The IETF RFC number for Session Description Protocol (SDP) bandwidth modifiers for RTCP bandwidth is wrong and is now corrected. Otherwise, the reference points to the wrong IETF document and this could result in problems for implementations.

Tdoc SP-030450: 26.976 “Performance characterization of the AMR-WB speech codec”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.976	001		Rel-5	Reference to incorrect test results	F	5.0.0	S4	TSG-SA WG4#28	S4-030625

- One of the references points to erroneous AMR-WB test results. This incorrect reference is replaced by a correct one. Otherwise, the reader when looking at the detailed test results (given in the reference) gets incorrect understanding of the performance of the AMR-WB codec in 8-PSK channels.

Tdoc SP-030451: 28.062 “Inband Tandem Free Operation (TFO) of speech codecs; Stage 3”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
28.062	040		Rel-5	Removal of Pre-Handover Notification for UMTS	F	5.3.0	S4	TSG-SA WG4#27	S4-030481

- Pre-Handover Notification option needs to be removed from 28.062 Annex D (TFO in 3G) since in UTRAN access, there is no need to steer the uplink and downlink rates into handover mode as the UE accepts any change within the ACS. If not removed, a procedure specified in the TFO specification cannot be implemented.

7. Miscellaneous

- A Low-Complexity AMR Noise Suppression (AMR-NS) solution from NEC Corporation was endorsed by SA4 at SA4#28. The endorsement means that, based on the test results presented to SA4, SA4 considers the algorithm meeting the recommended minimum performance requirements as given in 3GPP TS 26.077. A statement of this acknowledgement is included in the SA4#28 meeting report. (No AMR-NS algoritm itself is specified by SA4 nor standardised in 3GPP, i.e. the “endorsement” does not have such meaning. See TS 26.077 for details.)

8. Documents presented for information

The following documents are presented for information to TSG-SA from SA4:

- Tdoc SP-030434: Test and processing plan for default codec evaluation for speech enabled services (SES)
- Tdoc SP-030435: Test Plan for the AMR Narrow-Band Packet Switched Conversation test
- Tdoc SP-030436: Test Plan for the AMR Wide-Band Packet Switched Conversation test
- Tdoc SP-030437: AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan
- Tdoc SP-030438: PSS/MMS High-Rate Audio Selection Test and Processing Plan
- Tdoc SP-030439: Funding of Audio Codec Testing

9. Approval requested

SA4 requests TSG-SA#21 to approve the following:

SES codec recommendation criteria

Tdoc SP-030440: Recommendation Criteria for Default Codec for Speech Enabled Services (SES)

Work Item Descriptions:

Tdoc SP-030442: Updated WID on Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS) (Release 6)

New specifications:

Tdoc SP-030443: TR 26.937 on "RTP usage model" v. 2.0.0 (Release 5)

Change Requests:

Tdoc SP-030444: CR to TS 26.073 - Correction of the MMS_IO flag (Release 5)

Tdoc SP-030445: CR to TS 26.132 - Loudness rating measurements at lower bit rates (Release 5)

Tdoc SP-030446: CR to TS 26.173 - Possible decoder LPC coefficients overflow (Rel 5)

Tdoc SP-030447: CR to TS 26.204 - Possible decoder LPC coefficients overflow (Release 5)

Tdoc SP-030448: CRs to TS 26.234 - Corrections (Release 5)

Tdoc SP-030449: CRs to TS 26.236 - Corrections (Release 5)

Tdoc SP-030450: CR to TS 26.976 - Reference to incorrect test results (Release 5)

Tdoc SP-030451: CR to TS 28.062 - Removal of Pre-Handover Notification for UMTS (Release 5)

TSG-SA WG4 (SA4) - CODEC
Status Report at TSG-SA#21

Kari Järvinen
TSG-SA WG4 Chairman

☞ *SA4 status report in Tdoc SP-030433*

☞ *These slides in separate .ppt file included in Tdoc SP-030433 (Annex 1)*

Content

- **General issues**
- **Review of SA4 work progress**
  - **Release 5**
  - **Release 6**
- **Documents for information**
- **Documents for approval**

General: SA4 officials

- **Chairman:** **Kari Järvinen (Nokia / ETSI)**
- **Vice Chairmen:** **Catherine Quinquis (Orange, ETSI), Frédéric Gabin (NEC Technologies, ETSI)**
- **Secretary:** **Paolo Usai (3GPP Support)**
- **Sub Working Groups / ad-hocs:**
  - Speech Quality (SQ) Paolo Usai (ETSI)
  - Packet Switched Multimedia (PSM) Rolf Hakenberg (Panasonic / ETSI)
  - Audio codec ad-hoc group Imre Varga (Siemens / ETSI)

- **SA4 Vice Chairmen election took place at SA4#27 (in July)**

General: SA4 meetings

- **Meetings held**
  - **SA4#27** — **7 – 11 July, 2003** — **Host: Siemens; Venue: Munich, Germany**
  - **SA4#28** — **1 – 5 September, 2003** — **Host: Fraunhofer Institute; Venue: Erlangen, Germany**
- **Future meetings**
  - **SA4 Audio Codec Ad-Hoc (tbc)** — **7-9 October, 2003** — **Host: tbd; Venue: tbd**
  - **SA4 Video Codec Ad-Hoc** — **27- 29 October, 2003** — **Host: tbd; Venue: tbd**
  - **SA4#29** — **24 - 28 November, 2003** — **Host: Nokia; Venue: Tampere, Finland**
  - **SA4#30** — **23 - 27 February, 2004** — **Host: tbd; Venue: tbd**
  - **SA4#31** — **17 - 21 May, 2004** — **Host: tbd; Venue: tbd**
  - **SA4#32** — **16 - 20 August, 2004** — **Host: tbd; Venue: tbd**
  - **SA4#33** — **22 - 26 November, 2004** — **Host: tbd; Venue: tbd**
- **Meeting statistics**

Meeting	Number of (new) input documents	Number of participants	Number of incoming LSs	Number of outgoing LSs/communications
SA4#25	115	55	13	9
SA4#25bis	164	50	14	11
SA4#26	171	55	18	17
SA4#27	**142**	**65**	**19**	**14**
SA4#28	**128**	**55**	**18**	**9**

A GLOBAL INITIATIVE

General: Input documents

Tdoc	Title	Source	Agenda item	Document for
SP-030369	LS (from SA WG4) on "Assessment of the SES codecs ability to reconstruct speech"	SA WG4	7.4.1	Information
SP-030433	TSG S4 Status Report at TSG-SA#21	SA WG4 Chairman	7.4.1	Information
SP-030434	Test and processing plan for default codec evaluation for speech enabled services (SES)	SA WG4	7.4.1	Information
SP-030435	Test Plan for the AMR Narrow-Band Packet Switched Conversation test	SA WG4	7.4.1	Information
SP-030436	Test Plan for the AMR Wide-Band Packet Switched Conversation test	SA WG4	7.4.1	Information
SP-030437	AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan	SA WG4	7.4.1	Information
SP-030438	PSS/MMS High-Rate Audio Selection Test and Processing Plan	SA WG4	7.4.1	Information
SP-030439	Funding of Audio Codec Testing	SA WG4	7.4.1	Information
SP-030440	Recommendation Criteria for Default Codec for Speech Enabled Services (SES)	SA WG4	7.4.3	Approval
SP-030442	Updated Work Item Description on Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS) (Release 6)	SA WG4	7.4.3	Approval
SP-030443	TR 26.937 on "RTP usage model" v. 2.0.0 (Release 5)	SA WG4	7.4.3	Approval
SP-030444	CR to TS 26.073 - Correction of the MMS_IO flag (Release 5)	SA WG4	7.4.3	Approval
SP-030445	CR to TS 26.132 - Loudness rating measurements at lower bit rates (Release 5)	SA WG4	7.4.3	Approval
SP-030446	CR to TS 26.173 - Possible decoder LPC coefficients overflow (Rel 5)	SA WG4	7.4.3	Approval
SP-030447	CR to TS 26.204 - Possible decoder LPC coefficients overflow (Release 5)	SA WG4	7.4.3	Approval
SP-030448	CRs to TS 26.234 - Corrections (Release 5)	SA WG4	7.4.3	Approval
SP-030449	CRs to TS 26.236 - Corrections (Release 5)	SA WG4	7.4.3	Approval
SP-030450	CR to TS 26.976 - Reference to incorrect test results (Release 5)	SA WG4	7.4.3	Approval
SP-030451	CR to TS 28.062 - Removal of Pre-Handover Notification for UMTS (Release 5)	SA WG4	7.4.3	Approval

Release 5: Remaining Rel-5 work

- **TR 26.937 (RTP usage model) COMPLETED**
- **This "non-critical" TR brings additional information to characterise PS Streaming Service and gives useful information on issues that service providers and manufacturers should be aware of.**
- **Version 1.2.0 presented for information at TSG-SA#18 (in Tdoc SP-020683). Since then, updated versions have been prepared and feedback received from the relevant WGs (e.g., SA1, SA2, RAN2 and GERAN).**
- **The TR has now been finalised and is brought for approval (in Tdoc SP-030443).**

Release 6: SA4 WIDs

- **Performance Characterisation of Default Codecs for PS Conversational Multimedia Applications**
- **Packet Switched Streaming (PSS) Rel-6**
  - Non audio codec issues
  - Selection of default PSS/MMS audio codec
- **Extended AMR-WB codec ("AMR-WB+") Targeted for PS Streaming and Messaging Services**
- **Speech Recognition and Speech Enabled Services: Codec Work to Support Speech Recognition Framework for Automated Voice Services**
- **Media Codecs and Formats for IMS Messaging and Presence**
- **Definition of teleservice using MBMS**

Performance Characterisation of Default Codecs for PS Conversational Multimedia applications

- **The objective is to characterize the performance of default codecs for PS conversational multimedia applications (as defined in TS 26.235 "Default Codecs").**
- **Test Plans covering AMR narrowband and AMR wideband (AMR-WB) codecs finalised and are brought for information in Tdocs SP-030435 and SP-030436**
  - 24 test conditions both for the AMR and the AMR-WB codec
  - Error conditions with IP packet loss of 0% and 3%, and BER ratios of 10^{-2}, 10^{-3} and $5\ 10^{-4}$
  - Robust Header Compression (RoHC), an optional UMTS functionality, included for some test cases for AMR-WB tests
  - Three types of background noise: car, street and cafeteria
  - AMR codec modes 6.7 and 12.2 kbit/s, AMR-WB codec modes 12.65 and 15.85 kbit/s
  - The real-time test bed (UMTS simulator) provided by Siemens and France Telecom R&D
- **IPv6 is used in the characterisation**
  - IPv6 is simulated fully over the radio interface to ensure that the impact of bit-errors is properly taken into account for the IPv6 protocol
  - For the core network simulator, IPv4 is used instead of IPv6. However, as the core network simulator handles only packets (and not bits or bytes in packets), the use of IPv4 or IPv6 is not relevant in this part of the simulator: using IPv4 or IPv6 has no impact for characterisation purposes. The only impact is a minor difference in the bit-rate transmitted in the core network resulting into negligible difference in end-to-end delay. (The reason for using partly IPv4 is that the available core network simulator supports only IPv4.)

Performance Characterisation of Default Codecs for PS Conversational Multimedia applications

- **Subjective tests scheduled from mid-September until late November 2003**
  - ARCON, France Telecom R&D, and NTT-AT will act as subjective testing labs. Contracts for the involved laboratories have been finalised.
  - Budget "up to 194 kEuro" has been allocated earlier for this exercise (160 kEuro allocated by 3GPP PCG and the contingency of “up to 34 kEuro” left from the AMR-WB Characterisation Phase).
  - A compensation of 40 kEuro for each individual laboratory has been agreed, including testing and preparation of reports.
  - The contingency of maximum 34 kEuro will be utilized for global analysis of results, with the remaining part to be used for further characterisation testing (details to be decided in SA4).
  - A written confirmation by letter for authorizing the use of the AMR-WB contingency is still awaited from three companies.

- **Status of specifications**

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TR 26.9yz	Performance characterization of default codecs for PS conversational multimedia applications	SA4	-	TR preparation is pending on the testing.	TSG-SA#23 (March 2004)

Packet Switched Streaming (PSS) Rel-6

- **Rel-6 streaming has been discussed further with most debate on**
  - **bit-rate adaptation** (used to avoid quality degradations under critical conditions, e.g., due to network bandwidth variability and service interruptions such as handovers): SA4#28 agreed that no particular adaptation algorithm will be mandated (for PSS clients).
  - **quality metrics** (for servers to receive from the handset to provide the service providers means for evaluating the end user experience): some updates agreed e.g. for the RSTP protocol definition to send the quality metrics. SA4 internal "PSS Quality metrics" permanent document was updated accordingly (into v.0.06).
  - **consideration of new codecs**: Selection process continued for PSS/MMS audio codecs. Selection process planned for video codecs. GZIP chosen as the working assumption for compression of SVG (Scalable Vector Graphics) content; the compression allowing more efficient usage of network bandwidth.
- **Updated working drafts for TSs 26.234 (v.0.3.0), 26.244 (v.0.2.5), 26.245 (v.0.1.6) and 26.246 (v.0.0.3).**

Packet Switched Streaming (PSS) Rel-6

- **Enhanced video codecs discussed since SA4#27; selection phase planned.**
  - Characteristics of two codecs (MPEG-4 AVC/ITU-T H.264 and Microsoft WMV9) initially reviewed
  - Formal video codec selection process planned: preliminary version of candidate qualification criteria including preliminary time plan prepared (Annex 2 of this status report).
  - All proponents must notify via letter the SA4 Secretary of their intention to submit a candidate video codec by October 3rd 2003 (24:00 CET) in order to be considered in the selection process. (For planning purposes, it is necessary to know the maximum number of codecs at an early phase.)
  - SA4 Video Codec Ad-Hoc scheduled on 27-29 October (with decision power given by SA4 to finalise and approve the video codec candidate qualification criteria).
  - Candidate qualification material (as defined in the finalised qualification criteria document) to be submitted by each candidate for SA4#29 meeting in November.
  - The video codec selection is planned to take place at SA4#30 in February 2004.

- **Dialogue has been continued with OMA DLDRM working group regarding interworking issues between OMA DRM and 3GPP PSS Rel-6 service**
  - e.g., verifying suitability of OMA DRM v1.0 for progressive download in PSS.
  - SA3 has also been involved in the LS exchange between SA4 and OMA DLDRM, and SA4 understanding is that encryption will be done by SA3.

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Packet Switched Streaming (PSS) Rel-6

- **Status of specifications**

Deliverable	Title	Prime resp. WG	2nd resp. WG?	Comment/Status	TSG-SA approval target
CRs to TS 26.233	Transparent end-to-end PSS; General description	SA4	SA2	To be updated based on the content of PSS Rel-6.	TSG-SA#23 (March 2004)
CRs to TS 26.234	Transparent end-to-end PSS; Protocol and codecs	SA4	SA2	Working draft (v.0.3.0) exists in SA4	TSG-SA#23 (March 2004)
TS 26.244	Transparent end-to-end PSS; File Format	SA4	SA2	Working draft (v.0.2.5) exists in SA4	TSG-SA#23 (March 2004)
TS 26.245	Transparent end-to-end PSS; Timed Text Format	SA4	SA2	Working draft (v.0.1.6) exists in SA4	TSG-SA#23 (March 2004)
TS 26.246	Transparent end-to-end PSS; SMIL Language Profile	SA4	SA2	Working draft (v.0.0.3) exists in SA4	TSG-SA#23 (March 2004)
CRs to TS 22.233	Stage 1	SA1		Under SA1 responsibility.	
Possible new TS	Stage2 (non-transparent aspects)	SA2		To be produced by SA2, if needed.	

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)**

- **Test and processing plans finalised and are brought for information (in Tdocs SP-030437 and SP-030438)**

1) AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan

Experimental block A (Intrinsic quality comparison of candidate codecs)

- 14 kbps, mono, use case A (PSS)
- 18 kbps, stereo, use case A (PSS)
- 24 kbps, mono, use case A (PSS)
- 24 kbps, stereo, use case A (PSS)

Experimental block B (Quality comparison under stressed operating conditions)

- 14 kbps, mono, use case B (MMS), 16 kHz input and output sampling rate
- 18 kbps, stereo, use case B (MMS),
- 14 kbps, mono, use case A (PSS), 3% FER
- 24 kbps, stereo, use case A (PSS), 3% FER

2) PSS/MMS High-Rate Audio Selection Test and Processing Plan

Experiment 1: 32 kbps, stereo, use case A (PSS) and use case B (MMS)

Experiment 2: 48 kbps, stereo, use case A (PSS) and use case B (MMS)

Experiment 3: 32 kbps, stereo, use case A (PSS) with 1% and 3% random frame loss

*) This part reports jointly the audio codec work done in WIs "PSS Rel-6" and "Extended AMR-WB codec". (These WIs are related as AMR-WB+ codec is considered as one candidate for PSS/MMS default audio codec and testing will be carried out as combined testing.)

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- **8 listening laboratories worldwide**
  - T-Systems, NTT-AT, France Telecom R&D, Dynastat, Nokia, Ericsson, Coding Technologies, and Fraunhofer Institute. (Each test condition tested twice by two different laboratories.)
- **2 processing laboratories**
  - T-Systems and Audio Research Labs (each processing done twice for cross checking)
- **Audio material according to pre-defined generic audio signal categories**
  - A call is sent out for test material according to a pre-defined number of generic audio signal categories. Then, an independent selection entity will choose material representative for assumed typical applications to be used in the selection tests.
  - AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test contains four content types: Speech, Speech over Music, Speech between Music, and Music.
  - PSS/MMS High-Rate Audio Selection Test consists contains following categories: pop (with and/or without vocals), classic (with and/or without vocals), single instruments, a capella vocals (solo and/or choir), mixed speech and music, and speech with and/or without background noise.
- **Listening tests funded by the codec proponents (total of 487.5 kEuros)**
  - Funding of 400 kEuro for the AMR-WB+ and PSS/MMS Low-Rate tests, and 87.5 kEuro for the PSS/MMS High-Rate tests.
  - The funding is detailed in "Funding of Audio Codec Testing" document brought for information in Tdoc SP-030439.

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- **Candidate and reference codecs for AMR-WB+ and PSS/MMS Low-Rate testing**

#	Candidate Codec	AMR-WB+ candidate	PSS/MMS low-rate audio candidate	Supporting/providing Organization(s)
1	MPEG4 HE-AAC codec ("aacPlus")	No	Yes	Coding Technologies, NEC, Panasonic
2	AMR-WB+ candidate codec	Yes	Yes	Ericsson, Nokia, VoiceAge
3	CT codec ("Enhanced aacPlus")	No	Yes	Coding Technologies
#	**Reference codec**	-	-	**Providing Organization(s)**
5	AAC	-	-	Fraunhofer Institute
6	AMR-WB	-	-	3GPP

- **Candidate and reference codecs for PSS/MMS High-Rate testing**

#	Candidate Codec	Supporting/providing organization(s)
1	MPEG4 HE-AAC codec ("aacPlus")	Coding Technologies, NEC, Panasonic
2	CT codec ("Enhanced aacPlus")	Coding Technologies
#	**Reference codec**	**Providing Organization(s)**
3	AAC	Fraunhofer Institute
4	RealAudio	RealNetworks

RealAudio for information purposes only

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- The remaining permanent document on "PSS/MMS Audio Codec and Extended AMR-WB Selection Rules" has been progressed. (Draft version prepared based on methodology used for earlier codec selections in 3GPP e.g. for AMR-WB codec.)
- Could not be finalised and agreed at SA4#28 in time for TSG-SA#21 due to an outstanding issue in the format of the audio codec specification.
- SA4 has debated on how detailed description of the "winning codec" will be needed in the specifications, and specifically on
  - if the tested ANSI-C code of the PSS and MMS encoders will be included in the codec specifications; or
  - if minimum quality level could be guaranteed by defining normative minimum quality level specification (with the winner also obliged to make a reference source code as tested available outside specifications upon request under fair, reasonable and non-discriminatory terms and conditions).
- The following alternatives how to proceed were identified in extensive debate during SA4#28 (1-5 September) in the SA4 Audio Codec ad-hoc group…

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

PSS and MMS decoders:

- Specify decoder in each case by ANSI-C source code as integral and normative part of the specification and maintain it by usual 3GPP change control. Attach test vectors to be able to test compliance of implementations. AGREED

PSS encoder:

- **Method A**: Specify PSS encoder in each case by ANSI-C source code as integral part of the specification and maintain it by usual 3GPP change control. Attach test vectors to be able to test compliance of implementations.
- **Method B**: The normative specification of the encoder is a minimum quality level (description based on selection phase results). The meaning is that all Rel6 audio PSS compatible implementations shall not perform worse than this minimum quality level. No source code is attached. The winner is obliged to make a reference source code as tested, available upon request under fair, reasonable and non-discriminatory terms and conditions. Informative executable is attached. 3GPP SA4 is authorized to endorse implementations of the PSS encoder for compliance with the specification, according to a pre-defined test plan.
- **Method C**: require proponents to declare their policy as part of selection deliverables whether they follow PSS encoder Method A or Method B. Give preference in selection rules to proponents supporting Method A.
- **Method D**: require proponents to declare their policy as part of selection deliverables whether they follow PSS encoder Method A or Method B. Take the choice into account in selection rules**.**

MSS encoder:

(the same alternatives identified as for PSS encoder above)

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- **The approach for PSS and MMS decoders was agreed at SA4#28. However, no consensus between the "methods" for the PSS and MMS encoders.**
- **Further debated at SA4#28 with the following wording (for the Selection Rules) prepared as a potential compromise.**

"The format of the specification is as described below:

- Detailed technical description of the encoder and decoder

- ANSI-C source code of the tested encoder and decoder

Source code for both encoder and decoder (in the form as tested) shall be under the copyright and control of 3GPP (or one of its organizational partners) and shall be available from the controlling entity (3GPP or one of its organizational partners) to interested companies having a bona fide interest in receiving such source code.

Diligent efforts will be made to establish an infrastructure of confidential disclosure of the software, in order to ensure that valuable implementation details and trade secrets are not used by non-eligible entities, not used for products not compliant with the 3GPP specification and not used for non-3GPP applications. Access to the software may be subject to a reasonable fee paid to the controlling entity and will be granted solely for applications within the scope of the respective 3GPP specification. (Wording to be reviewed by 3GPP legal adviser.)"

- **Tdoc S4-030703 "Draft PSS/MMS Audio Codec and Extended AMR-WB, Selection Rules Version 0.4" (Annex 3 of this report) with the above text, put for e-mail approval after SA4#28 until September 12th.**

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- **Two objections (from France Telecom / Orange and Nokia), both on the proposed limited access to the source code.**
  - France Telecom / Orange pointed out two issues: 1) A definition of "non-eligible entity" (or of "eligible entity") is necessary; and 2) What is the purpose of the fee?
  - Some discussion on these issues followed on the SA4 reflector, but the France Telecom / Orange objection was sustained, claiming an apparent contradiction between the fact that ETSI cannot discriminate (to whom to deliver the source code) and that there could be non-eligible entities.
  - The Legal Adviser at ETSI, Mr. Stephane Tronchon, stated (consulted by SA4 Secretary) that the text of SA4 document is basically fine 'as is', but that it needs to be interpreted taking into account further explanations/clarifications and existing similar cases where ETSI is acting as the Custodian of source codes.
  - As a result of the discussion on SA4 reflector and the raised interpretations of eligible and non-eligible entities, also Nokia expressed that they do not accept the text.
  - The Selection Rules were thus not approved by SA4 and therefore could not be finalised by TSG-SA#21 as planned.
- **Audio codec ad-hoc meeting scheduled for October 7-9 (tbc) in case Selection Rules not approved by correspondence (which happened). If agreement on specification format not reached at/by the audio codec ad-hoc meeting, it could at the worst jeopardise Rel-6 schedule for the PSS/MMS audio codec work.**
- **Subjective testing scheduled for December-January timeframe with first steps for the testing to be taken already during October (submission of candidate codecs and preparation of the test material and error patterns for the testing).**
- **Advice is sought from TSG-SA on the above matter remaining unsolved in SA4!**

Audio codecs *(PSS/MMS default audio codec, extended AMR-WB codec)*

- **Status of specifications**

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
CRs to 26-series AMR-WB TSs/TRs	(Relevant AMR-WB specifications of 26 –series)	SA4	-	Drafts to be prepared by candidates for the codec selection meeting (SA4#30).	TSG-SA#23 (March 2004)
New audio codec TS(s)		SA4	-	Drafts to be prepared by the candidates for the codec selection meeting (SA4#30).	TSG-SA#23 (March 2004)
CRs to TS 26.234	Transparent end-to-end PSS; Protocols and codecs	SA4	SA2	Default codec definition for the audio media type needs to be updated based on the selection of new audio codec(s).	TSG-SA#23 (March 2004)

Codec Work to Support Speech Recognition Framework for Automated Voice Services

- **Two SES default codec candidates: 1) DSR AFE (ETSI DSR standard ES 202 050) and its extension and 2) the AMR and AMR-WB codecs**
- **The remaining permanent document on Recommendation Criteria completed in Tdoc SP-030440**
  - Provides criteria for making recommendation for SES default codec.
  - First, any candidates not compliant with all design constraints excluded.
  - Then, the candidates will be compared against each other based on relative reduction in average word error rate.
- **Test and Processing Plan is presented for information in Tdoc SP-030434**
  - Tasks: connected digit recognition task, sub-word trained model recognition task and tone confusability task.
  - Testing is done in error-free channel as well as under packet loss situations. The channel error experiments cover average channel BLock Error Rates (BLER) of 1% and 3%. In addition, a BLER of 10% is tested for informative purposes. The testing covers Error Patterns (EP) for UTRAN and EGPRS/GPRS channels.
  - AMR modes of 4.75 and 12.2 are included in the tests for 8 kHz sampling rate test case, and AMR-WB modes of 12.65 and 23.85 for 16 kHz sampling rate case.
  - Two Automatic Speech Recognition (ASR) vendors, IBM and SpeechWorks (now ScanSoft), will carry out the testing (on voluntary basis). Preparation for the tests is currently ongoing. Both vendors run tests for both candidates.
- **The results of evaluations reported to SA4#29 (November) for making recommendation. The codec selection for approval at TSG-SA#22 (December).**

Codec Work to Support Speech Recognition Framework for Automated Voice Services

- **TSG-SA#20 requested SA4 to assess the codecs' ability to reconstruct speech.**
  - SA4 recognizes that both SES codec candidates are capable of reconstructing intelligible speech (on grounds of existing DSR test results from ETSI Aurora and existing AMR/AMR-WB test results in 3GPP). Reconstruction quality of codec candidates will be measured by interested SA4 companies but for informative purposes only. A LS on this is presented in Tdoc SP-030369.

- **SA4 responded to LS from SA2 on "Usage of Speech Enabled services in CS Domain"**
  - SA4 has no data available currently to quantify the potential improvement by the mentioned methods for CS domain (indication of codec type for the speech recogniser, introduction of specific speech recognition mode in the terminal), and this would require further investigation which is not currently on-going in SA4.
  - SA4 is working on SES codecs for PS domain. SA4 will keep SA2 informed on the results of this work if relevant for the speech recognition performance in CS domain.

- **Status of specifications**

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
CRs to TS 26.235	PS Conversational Multimedia Applications; Default Codecs	SA4	SA2, T2	To be prepared based on the codec selection.	At the earliest at TSG-SA#22 (Dec 2003)
CRs to TS 26.236	PS Conversational Multimedia Applications; Transport Protocols	SA4	SA2, T2	To be prepared based on the codec selection	At the earliest at TSG-SA#22 (Dec 2003)
Possible new TSs	Codec specification	SA4		To be prepared, if needed.	At the earliest at TSG-SA#22 (Dec 2003)

Media Codecs and Formats for IMS Messaging and Presence

- **A first "skeleton" working draft of TS 26.141 "IMS Messaging and Presence; Media formats and codecs" has been prepared.**
  - Based on using the legacy media types (and codecs) from MMS as starting point. The discussion of new audio and video codecs for PSS/MMS in Rel-6 ongoing in SA4 will be taken into account.
  - The emerging Common Presence and Instant Messaging Message format CPIM (details being speficied by CN1 in TS 24.841) and the technical requirements following from this are to be considered within the SA4 work.

- **Status of specifications**

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TS 26.141	IMS Messaging and Presence; Media formats and codecs	SA4	SA2, CN1	First skeleton working draft prepared.	TSG-SA#23 (March 2004)

Definition of teleservice using MBMS

- Preparation of TS on "MBMS Protocols and Codecs TS" started in SA4 with an initial "skeleton" draft; the work guided by the ongoing SA1 requirements work. (A draft SA1 TS on Stage 1 on "MBMS User Services" from SA1 presented for information at SA4#28.)
- SA1 explained that a full definition of MBMS teleservice is unnecessary to fulfil the targets of the work, and consequently, asked SA4 to modify the WID to reflect this (with suggested revisions). SA4 agreed and is bringing a revised WID for approval in Tdoc SP-030442.
- A joint meeting between SA1, SA2 and the involved RAN and GERAN WGs might be beneficial. (Relevant WG Chairmen will be contacted off-line to find out if there is any interest or possibilities for such a joint meeting.)
- Status of specifications

Deliverable	Title	Prime resp. WG	2nd resp. WG	Comment/Status	TSG-SA approval target
TS 26.x.y.z	MBMS Protocols and Codecs	SA4	SA2, SA3	First skeleton working draft prepared.	TSG-SA#23 (March 2004)

Miscellaneous

- **A Low-Complexity AMR Noise Suppression (AMR-NS) solution from NEC Corporation was endorsed by SA4 at SA4#28.**
  - The endorsement means that, based on the test results presented to SA4, SA4 considers the algorithm meeting the recommended minimum performance requirements as given in 3GPP TS 26.077.
  - A statement of this acknowledgement is included in the SA4#28 meeting report. (No AMR-NS algoritm itself is specified by SA4 nor standardised in 3GPP, i.e. the “endorsement” does not have such meaning. See TS 26.077 for details.)

Communication with other WGs/TSGs/groups

Tdoc no.	Title	Intended for	Copy to
TD S4-030512	(Reply) LS on clarification on minimum set of TFCs for TFC selection	RAN2	RAN4
TD S4-030549	Liaison Statement on Discard Timer	RAN3	SA2, RAN2, GERAN2
TD S4-030550	Liaison response on Meta-Data in ISO Media Files, Streaming Text, Advanced Text and Graphics Amendment	ISO/IEC SC29/WG11	
TD S4-030510	LS on DRM for Progressive Download	OMA MAG/DL+DRM	SA3
TD S4-030518	Reply to "Liaison regarding RTP timestamps"	IMTC PSS AG	
TD S4-030509	Communication / LS on the specified levels in ITU-T Recommendation H.264 | ISO/IEC International Standard 14496-10 (MPEG-4 AVC)	ITU-T SG16 WP3 Question 6 and Joint Video Team; ISO/IEC JTC1/SC29/WG11 (MPEG)	
TD S4-030529	Liaison Statement on specified levels in MPEG-4 Visual Simple Profile	ISO/IEC JTC1/SC29/WG11 (MPEG)	
TD S4-030532	LS on compression of SVG content and progressive downloading	W3C SVG group	
TD S4-030547	LS on SA4 assessment of the SES Codecs ability to reconstruct speech	TSG SA	
TD S4-030558	Reply to LS on Core Network Provision of separate flows for P2P and P2M radio Transmission	CN1, CN4, RAN1, RAN2, GERAN1, GERAN2, SA1, SA2	
TD S4-030573	Reply to LS on future codecs with same SDU format as existing codecs	TSG CN, CN4	
TD S4-030557	Communication to ITU-T on the specified levels in Annex X of ITU-T Recommendation H.263	ITU-T SG16 WP3 Question 6 (Video Coding Experts Group)	
TD S4-030552	LS on DRM issues	OMA MAG/DL+DRM	
TD S4-030576	LS on DRM issues	ISMA	
TD S4-030660	LS on cipher suite for DRM-protected streamed media for PSS	SA3	
TD S4-030647	Liaison Response to OMA on DRM issues	OMA	SA3
TD S4-030683	Communication to ITU-T SG9 on Timed Text	ITU-T SG9	
TD S4-030684	Liaison to 3GPP2 on Timed Text	3GPP2	
TD S4-030670	LS on "Update of WID on MBMS"	SA1	SA2, SA3, SA5, RAN2, RAN3, GERAN1, GERAN2, CN1
TD S4-030685	Response to Liaison Statement on "Reliable transport" for PSS	SA1	
TD S4-030686	Reply to LS on "Usage of RTCP & SDP in MBMS"	RAN2	SA2, RAN3, GERAN1, GERAN2
TD S4-030679	Reply to "Usage of Speech Enabled Services in CS Domain"	SA2	
TD S4-030687	LS on scalable codecs for MBMS	RAN2	RAN3, SA2

Documents for information

- **Tdoc SP-030434: Test and processing plan for default codec evaluation for speech enabled services (SES)**
- **Tdoc SP-030435: Test Plan for the AMR Narrow-Band Packet Switched Conversation test**
- **Tdoc SP-030436: Test Plan for the AMR Wide-Band Packet Switched Conversation test**
- **Tdoc SP-030437: AMR-WB+ and PSS/MMS Low-Rate Audio Selection Test and Processing Plan**
- **Tdoc SP-030438: PSS/MMS High-Rate Audio Selection Test and Processing Plan**
- **Tdoc SP-030439: Funding of Audio Codec Testing**

Documents for approval

- **Tdoc SP-030440: Recommendation Criteria for Default Codec for Speech Enabled Services (SES)**
  - Candidates not compliant with all design constraints are first excluded. Then, remaining candidates will be compared against each other based on relative reduction in average word error rate in selection test results.
  - The AMR/AMR-WB codec mode used is AMR 4.75, AMR 12.2, and AMR-WB 12.65. In case the relative reduction for the DSR AFE codec and its extension compared to the AMR 4.75 / AMR 12.2 / AMR-WB 12.65 codec is more than 35% / 30% / 25%, respectively, then the DSR AFE codec and its extension will be recommended.
  - In case the reduction is less than 20% / 20% / 15% then the AMR/AMR-WB codec will be recommended.
  - With results in between these limits (i.e., 20% and 35%, 20% and 30%, and 15% and 25%), the performance results will be further considered by SA4 (and if no consensus the results will be passed to TSG-SA for decision on what recommendation to make).
  - This procedure for analysing the relative reduction in average word error rate is carried out for low-rate data comparison at 8 kHz sampling rate and for high data-rate comparison at 8 kHz and at 16 kHz.

Documents for approval: updated WID

- **Tdoc SP-030442: Updated WID on Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS) (Release 6)**
  - The joint SA4/SA1 WID on “Definition of teleservice using MBMS” approved at TSG-SA#20 has been updated into “Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS)”
  - SA1 explained that a full definition of MBMS teleservice is unnecessary to fulfil the targets of the work, and consequently, asked SA4 to modify the WID to reflect this (with suggested revisions).

Documents for approval: new TR

- **Tdoc SP-030443: TR 26.937 on "RTP usage model" v. 2.0.0 (Release 5)**
  - This "non-critical" TR brings additional information to characterise PSS (e.g., statistics of traffic characteristics such as packet sizes and bit-rates) and gives useful information on issues that service providers and manufacturers should be aware of (e.g., implications of chosen RTP packet sizes and impact of different rate control strategies for video streaming).
  - The document discusses, e.g., the following issues:
    - Trade-off between radio usage efficiency and streaming QoS
    - Feedback of network conditions and adaptation of stream and/or the transmission of the stream
    - Optimal packetisation of the media stream in line with the segmentation within the transport mechanism
    - Error robustness mechanisms (such as retransmission)
    - Client buffering to ease the QoS requirements on the network and enable more flexibility in how the network transport resources are applied
  - Version 1.2.0 presented for information at TGS-SA#18 in Tdoc SP-020683.
  - Draft versions reviewed by the relevant WGs (SA1, SA2, RAN2 and GERAN). Feedback taken into account.

Documents for approval: Change Requests

Tdoc SP-030444: 26.073 "ANSI-C code for the Adaptive Multi Rate (AMR) speech codec"

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.073	018		Rel-5	Correction of the MMS_IO flag	F	5.1.0	S4	TSG-SA WG4#27	S4-030485

- The code of the AMR codec does not compile when the MMS option is enabled. Therefore, the developers can not test the code with the AMR MIME file storage format unless the bug is corrected.

Tdoc SP-030445: 26.132 "Speech and video telephony terminal acoustic test specification"

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.132	026		Rel-5	Loudness rating measurements at lower bit rates	F	5.3.0	S4	TSG-SA WG4#28	S4-030619

- A note is added to explain that the use of multisine signal is not recommended for measurements of loudness ratings at lower AMR bit-rates than 12.2 kbit/s. Although only 12.2 kbit/s mode is defined for use in testing, some manufacturers have done additional testing with the lower bit-rate modes and come up with peculiar results. (For multisine signal, the test results depend very much on the selected AMR bit-rate.)

Tdoc SP-030446: 26.173 "ANSI-C code for the AMR-WB codec"

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.173	019		Rel-5	Possible decoder LPC coefficients overflow	F	5.7.1	S4	TSG-SA WG4#28	S4-030634

- AMR-WB decoder can produce unstable output during DTX-operation (source controlled operation). Conversion from ISP to LPC coefficients is changed so that LPC coefficients cannot overflow in decoder comfort noise generation. Otherwise, the decoder synthesis filter may become unstable causing uncontrolled output when DTX is used.

Tdoc SP-030447: 26.204 "ANSI-C code for the Floating-point AMR-WB codec"

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.204	008		Rel-5	Possible decoder LPC coefficients overflow	F	5.1.0	S4	TSG-SA WG4#28	S4-030635

- (same reason as above)

Documents for approval: Change Requests

Tdoc SP-030448: 26.234 “Transparent end-to-end PSS; Protocols and codecs“

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.234	061	1	Rel-5	Clarification on session bandwidth for RS and RR RTCP modifiers	F	5.5.0	S4	TSG-SA WG4#27	S4-030556
26.234	062	1	Rel-5	Correction of ambiguous range headers in SDP	F	5.5.0	S4	TSG-SA WG4#27	S4-030511
26.234	063	1	Rel-5	Timed-Text layout example	F	5.5.0	S4	TSG-SA WG4#27	S4-030555
26.234	064		Rel-5	Correction of ambiguity in RTP timestamps handling after PAUSE/PLAY RTSP requests	F	5.5.0	S4	TSG-SA WG4#27	S4-030517
26.234	065		Rel-5	Correction of obsolete RTP references	F	5.5.0	S4	TSG-SA WG4#28	S4-030607
26.234	066	1	Rel-5	Correction of wrong reference	F	5.5.0	S4	TSG-SA WG4#28	S4-030648
26.234	067		Rel-5	Missing signaling of live content	F	5.5.0	S4	TSG-SA WG4#28	S4-030654

- These contain the following corrections:
  - Session bandwidth for RS and RR RTCP modifiers are corrected to avoid misinterpretation. Otherwise, the rules are ambiguous and interoperability problems will occur.
  - The overview table of SDP fields indicates that the a=range field is required on both session and media levels, which is inconsistent with the usage of the field explained elsewhere. A note is added explaining the correct usage. Otherwise, media streams of different lengths would be ambiguous and were interpreted differently by different clients thus impacting interoperability.
  - The Timed-Text layout example given is wrong. The character order is corrected. Otherwise, implementations may follow the example resulting in incomprehensible text. (Even though it is an example, it is an important part of the specification. The text is hard (if not impossible) to understand without a correct example.)
  - Interoperability tests have showed that the current specification is unclear about how a PSS server has to timestamp RTP packets after a PLAY request. Text is added clarifying what must be done by the server with RTP timestamps. Otherwise, different interpretations would lead to non-interoperable solutions and prevent the playback control to work properly.
  - Obsolete RTP references have been corrected Otherwise, the mandatory SDP bandwidth modifiers RR and RS cannot be correctly implemented and implementers will encounter problems.
  - The IETF RFC number for Session Description Protocol (SDP) bandwidth modifiers for RTCP bandwidth is wrong and is now corrected. Otherwise, the reference points to the wrong IETF document and this could result in problems for implementations.
  - There are two types of media a PSS server can offer: on-demand and live. In the current PSS specification, there is no explanation on how the server can signal live content and therefore a client will not be able to act accordingly. Explanation on how live content can be signalled is included. Otherwise, a client will not be aware that the offered content is live, which may lead to interoperability problems and failure of service.

Documents for approval: Change Requests

Tdoc SP-030449: 26.236 “PS conversational multimedia applications; Transport protocols“

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	006		Rel-5	Correction of obsolete RTP references	F	5.3.0	S4	TSG-SA WG4#28	S4-030608
26.236	007	1	Rel-5	Correction of wrong reference	F	5.3.0	S4	TSG-SA WG4#28	S4-030649

- Obsolete RTP references have been corrected Otherwise, the mandatory SDP bandwidth modifiers RR and RS cannot be correctly implemented and implementers will have problems resolving ambiguities.
- The IETF RFC number for Session Description Protocol (SDP) bandwidth modifiers for RTCP bandwidth is wrong and is now corrected. Otherwise, the reference points to the wrong IETF document and this could result in problems for implementations.

Tdoc SP-030450: 26.976 “Performance characterization of the AMR-WB speech codec”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.976	001		Rel-5	Reference to incorrect test results	F	5.0.0	S4	TSG-SA WG4#28	S4-030625

- One of the references points to erroneous AMR-WB test results. This incorrect reference is replaced by a correct one. Otherwise, the reader when looking at the detailed test results (given in the reference) gets incorrect understanding of the performance of the AMR-WB codec in 8-PSK channels.

Tdoc SP-030451: 28.062 “Inband Tandem Free Operation (TFO) of speech codecs; Stage 3”

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
28.062	040		Rel-5	Removal of Pre-Handover Notification for UMTS	F	5.3.0	S4	TSG-SA WG4#27	S4-030481

- Pre-Handover Notification option needs to be removed from 28.062 Annex D (TFO in 3G) since in UTRAN access, there is no need to steer the uplink and downlink rates into handover mode as the UE accepts any change within the ACS. If not removed, a procedure specified in the TFO specification cannot be implemented.

(end of presentation)

Source: PSM SWG

Title: **Advanced video codec selection: Candidate requirements.**

Agenda Item: 6.6.5

Document for: Approval

1 Introduction

This document reflects work in progress. All items here are subject to change. Comments & corrections welcome.

3GPP Release 6 requires an advanced video codec that achieves a significantly better user experience at 32, 64, 128 and 256 kbps than is achieved with Release 5 video codecs. This codec is expected to be used for MMS, PSS and PSC services.

This document describes the process to identify candidates. Other related documents are (S4-030xxx and S4-030xxx) which describe the criteria by which a winner will be selected, and the test plan to evaluate those criteria.

This document contains the following sections:

1. This introduction.
2. Description of deliverables and other requirements to enable the standardisation and implementation of a candidate codec in 3GPP services.
3. Description of technical criteria that a proposed codec must meet to qualify as a candidate.
4. Description of items that must be submitted for a proposed codec to SA4 so that its qualification as a candidate can be assessed.
5. Time plan detailing meeting dates and critical steps in the selection process.

1.1 Glossary

Item	**Description**
The Reference	The Existing Release 5 codec that the candidate or potential candidate is compared with. The Reference is: xxxx (ed: what is it?)

2 "The Reference" The Deliverables and other requirements for standardisation.

The following must be publicly available for the successful candidate to become a 3GPP standard. The Candidates must commit, at the point of notifying their candidature, to meet these requirements. Should a candidate be successful, and this obligation not be fulfilled to the satisfaction of SA4 in the February 2004 meeting, then it will be disqualified.

- A decoder specification for the candidate codec in form of a 3GPP TS or as a reference to a public document.

- An RTP payload format specification for the candidate codec that can be used in PSS.
- A storage format mapping so that the candidate codec may be contained in the 3GP file format for MMS
- Details of required media specific SDP parameters including MIME type specification.
- Reference decoder source code – that facilitates understanding of the specification and aids compatibility of implementations. Shall be informatively referenced by the 3GPP specification.
- Reference encoder source code -that facilitates understanding of the specification and aids compatibility of implementations. Shall be informatively referenced by the 3GPP specification.

In addition, IP owners for candidate codec must commit to comply with 3GPP IPR rules.

3 Technical criteria for candidacy.

3.1 A candidate must achieve the same visual quality as The Reference at a significantly lower bit rate.

- Proposed candidates must have the same or better PSNR than the reference has when encoded at a 50% higher bit rate, averaged over the entire clip. This must be achieved for all clips at the following settings:
  - QCIF 15fps: 48kbps or less for candidate, 72kbps for reference
  - QCIF 15fps: 96kbps or less for candidate, 144kbps for reference
- Clips will be (music video, movie trailer, news, football, traffic, foreman).
- Uncompressed AVI sources and reconstructed reference AVI files are available from: XXXXX (Ed: where?)

3.2 A Candidate must consume no more than 3 times the average CPU and 5 times the maximum memory that the reference uses for any clip.

- Decoder performance reported must use all decoder options necessary to achieve the PSNR reported for 3.1 above.
- Average CPU consumption means the average Megacycles per second over the whole clip.
- Figures reported may exclude rendering and colour conversion.
- Measurement must be against same encoded clips used in 3.1 above.
- Measurement must be on Armulator, emulating Arm920T.
- The reference average CPU and maximum Memory figures for the clips in the test environment are:

Clip	Memory	Mcps

4 Candidature qualification submission requirements.

A proposed candidate must submit the following items to prove that it qualifies:

- Candidate criteria pass documentation.
    - Details of PSNR figures for all clips and bit rates indicated in 3.1 above
    - Details of MCPS and memory consumed for all clips and bit rates as described .in 3.2 above
- Decoded output in YUV format with same number of frames as source material.
- Encoded media files.
- Tools to decode and convert encoded media to uncompressed YUV.

5 Time plan

5.1 Proponents must notify 3GPP of intention to submit a candidate by October 3 2003 (24:00 CET)

5.1.1 Submission of intent must be sent to Paolo Usai (…..)

5.2 Candidature qualification material must be submitted by November 19th 2003 (24:00 CET)

5.2.1 Submission must be in form of 3GPP SA4 input document.

(Ed note: Need to acquire right to finalise candidate qualification criteria document in October meeting.)

September	Today	SA4#28 (1-5 Sept)	TSG-SA#21 (22-25 Sept.)
October	Declaration of intention to submit a candidate (all candidates have to make this declaration to be considered in this process) Candidate qualification criteria document finalised Start to work on other documents	03 Oct Video ad-hoc (week starting	

		from 27 Oct, three days)	
November	Submission of video codec candidates Selection of candidates Test plan document finalised Work on selection criteria document, finalise if possible	SA4#29 (24-28 Nov)	
December	Selection criteria document finalised, if not done at SA4#29	Video ad-hoc (week starting from 15 Dec, three days)	TSG-SA#22 (15-18 Dec)
2004 January	All test results are available Review of test results	Video ad-hoc (week starting from 19 Jan, three days)	
February	Selection of video codec(s) All decisions need to be taken Final specifications should be available	SA4#30 (23-27 Feb)	
March			TSG-SA#23 (15-17 March)
April			
May	Verification of specifications (if needed) Finalisation of non-critical specifications	SA4#31 (17-21 May)	

Source: Audio codec ad-hoc

Title: Draft PSS/MMS Audio Codec and Extended AMR-WB, Selection Rules Version 0.4~~1~~

Introduction

This document contains a proposal for a permanent document on Selection Rules for PSS/MMS Audio Codec and AMR-WB+. This document was prepared based on permanent document on Selection Rules used earlier in 3GPP for AMR-WB codec selection. (Since no separate permanent document exists this time for Selection Deliverables, a list of required selection deliverables is included in Annex A of this proposed permanent document.)

For permanent documents of AMR-WB Selection Phase, see http://www.3gpp.org/ftp/tsg_sa/WG4_CODEC/AMR-Wideband/Perm_Docs_Selection_Phase.

1. PSS/MMS Low-Bit Rate Audio Codec (LBRAC) Selection Rules

Three basic rules are defined. The first two rules are eliminating rules intended to exclude all candidates failing to demonstrate full compliance with the PSS/MMS Audio Codec Design Constraints defined in [1] or presenting test results too far below the expected performance level. The third rule is not exactly a rule but a primary selection of Figures of Merit according to which the candidate performances will be compared as part of the Selection test results analysis. These multiple criteria are intended to provide a good picture of the relative performances of the proposed solutions.

Each rule is further described in the following sections:

PSS/MMS LBRAC Selection Rule 1:

Any candidate (including AMR-WB+) not compliant with all Design Constraints defined in PSS/MMS Audio Codec Design Constraints permanent document [1] will be excluded. In the case when the AMR-WB+ candidate fulfils the PSS/MMS audio codec design constraints and wins the selection based on the rules defined in this document but fails to fulfil the AMR-WB+ design constraints, the adoption of AMR-WB+ codec as the default PSS/MMS low bit-rate audio codec will be determined in TSG-SA4 group.

PSS/MMS LBRAC Selection Rule 2:

Any candidate not meeting the performance requirements will be excluded. In order to meet the performance requirements, a candidate must be better than the reference at least in one ~~experiment~~test case (a test case being defined by its bitrate and mono/stereo configuration). A candidate must never be worse than the reference in any ~~experiment~~ test case in experimental block A. However, up to one failure is accepted in quality comparison under stressed operating conditions (experimental block B).). In each test case independently, the reference is selected as either AMR-WB or MPEG-4 AAC LC whichever performs better on the average over all tested items.

Denoting the AAC-LC performance in test case K and for item i (N is the total number of tested items) by $P_{AAC}(K,i)$, and respectively, the AMR-WB performance by $P_{AMRWB}(K,i)$, then the reference performance is understood as:

$$R(K) = \max\left(\tfrac{1}{N}\sum_{i=1}^{N} P_{AAC}(K,i), \tfrac{1}{N}\sum_{i=1}^{N} P_{AMRWB}(K,i)\right).$$

~~Reference is understood as the better of AMR-WB and MPEG-4 AAC LC at the same bit-rate in any test case based on the average performance over music, mixed content and speech.~~ According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.

~~[Note: Is the reference performance achieved by:~~

~~-Maximum score of the two codecs averaged over all items~~

~~-Average of the maximum score of the two codecs averaged over items in each content type~~

~~-Average of the maximum score of the two codecs in each item]~~

~~The performance of the codec under test is equal to the average score over all items.~~

"Better than" and "no worse than" are defined at the 95% confidence level for performance measures defined above.

The score is understood as a MUSHRA score averaged across the replications of the sub-experiments (different laboratories) in each operational mode and operational condition.[1]

PSS/MMS LBRAC Selection Rule 3: Figures of Merit:

A number of Figures of Merit will be used to analyse and compare the performance of the candidates. Corresponding rankings will be prepared and provided for information only. None of the Figures of Merit listed below is intended to serve as single selection criteria.

The candidates will be ranked according to the following metrics:

Preferred quality FoM:

For each test case *K* and content type *T* = {music, speech over music, speech between music, speech} a delta performance score is calculated as the difference between the codec performance $\overline{P}_C(K,T)$ and the performance of a quality reference $R_Q(K,T)$:

$$\Delta_C(K,T) = \overline{P}_C(K,T) - R_Q(K,T)$$

The quality reference is calculated according to:

$$R_Q(K,T) = \frac{1}{N_T}\sum_{i\in I_T} \max(P_{AAC}(K,i), P_{AMRWB}(K,i)),$$

where I_T denotes the set of N_T items belonging to content type *T*.

The candidate codec performance is calculated according to:

$$\overline{P}_C(K,T) = \frac{1}{N_T}\sum_{i\in I_T} P_C(K,i)$$

The delta performance scores are arranged in a matrix where the different content types are given across the columns while the test cases are across the rows. Negative delta scores will be highlighted in red in order to indicate cases where the reference performance it is not met.

In addition to the item-wise delta scores per test cases, average, minimum and maximum delta scores will be given both across content type and test case.

The minimum and maximum delta score is understood as worst, respectively, best observed score across all items separately in content types and in test cases in which the reference is taken as maximum of the performances of AAC-LC and AMR-WB for the respective items at the given content types and test cases.

In order to provide a global overview further composite scores are derived such as average, minimum and maximum scores across the complete set of test cases and content types.

An overview of the complete matrix of scores is given in table 1:

Content type Operating	Music	Speech over music	Speech between music	Speech	Average	Min per item	Max per item

[1] According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.

condition							
14 kbps, mono, use case A (PSS)							
18 kbps, stereo, use case A (PSS)							
24 kbps, mono, use case A (PSS)							
24 kbps, stereo, use case A (PSS)							
14 kbps, mono, use case B (MMS), 16 kHz inp. and outp. sampling rate							
18 kbps, stereo, use case B (MMS)							
14 kbps, mono, use case A (PSS), 3% FER							
24 kbps, stereo, use case A (PSS), 3% FER							
Average							
Min per item							Not used
Max per item						Not used	

~~A number of Figures of Merit will be used to analyse and compare the performance of the candidates. Corresponding rankings will be prepared and provided for information only. None of the Figures of Merit listed below is intended to serve as single selection criteria. A preferred criterion for the quality assessment of the candidates should be defined among the Figures of Merits, if possible.~~

~~The candidates will be ranked according to the following metrics:~~

~~*FoM 1a:*~~

~~Sum of delta MUSHRA against the better of AMR-WB and AAC LC reference codec over all test items. According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.~~

FoM L1~~1b~~:

The number of positive delta MUSHRA table entries. ~~items where the candidate performs better than the better of AMR-WB and AAC LC reference codec. According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.~~

~~*FoM 1c:*~~

~~Sum of delta MUSHRA of items where the candidate performs better than the better of AMR-WB and AAC LC reference codec. According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.~~

FoM L2~~1d~~:

The number of negative delta MUSHRA table entries. ~~items where the candidate performs worse than the better of AMR-WB and AAC LC reference codec. According to the content type weighting specified in [1],~~

~~the experimental results for mixed content (speech over music and speech between music) are counted twice.~~

~~*FoM 1e:*~~

~~Sum of delta MUSHRA of items where the candidate performs worse than the better of AMR-WB and AAC LC reference codec. According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.~~

~~*FoM 2-11:*~~

~~Weighted Delta MUSHRA against the better of AMR-WB and AAC LC reference codec over each test item separately~~[2]

~~List of low bit rate test sets (FoM):~~
~~Set #1: 14 kbit/s mono (FoM 2)~~
~~Set #2: 14 kbit/s mono (use case B, 16 kHz) (FoM 3)~~
~~Set #3: 18 kbit/s stereo (FoM 4)~~
~~Set #4: 18 kbit/s stereo (use case B) (FoM 5)~~
~~Set #5: 24 kbit/s mono (FoM 6)~~
~~Set #6: 24 kbit/s stereo (FoM 7)~~
~~Set #7: speech content (FoM 8)~~
~~Set #8: speech over music content (FoM 9)~~
~~Set #9: speech in between music content (FoM 10)~~
~~Set #10: music content (FoM 11)~~
~~Set #11: channel impairments (FoM 12)~~

~~*FoM 13-88:*~~

~~Minimum Delta MUSHRA against the better of AMR-WB and AAC LC reference codec across all items within each content class.~~

~~Maximum Delta MUSHRA against the better of AMR-WB and AAC LC reference codec across all items within each content class.~~

FoM L3a-L3e~~89-N~~:

Figure of Merit for computational complexity and memory are: ~~(TBA)~~

- The peak-WMOPS (measured for the worst observed frame)
- Average-WMOPS (measured over the test material)
- RAM (in kWords measured for the worst test case)
- ROM (in kWords measured for the worst test case)
- Program ROM (number of instructions measured for the worst test case)

~~[Note: FoM is needed for design objectives.]~~

[2] According to the content type weighting specified in [1], the experimental results for mixed content (speech over music and speech between music) are counted twice.

2 Set of Rules for High-Bit Rate Audio Codec (HBRAC) Selection Rules

Three basic rules are defined. The first two rules are eliminating rules intended to exclude all candidates failing to demonstrate full compliance with the PSS/MMS Audio Codec Design Constraints defined in [1] or presenting test results too far below the expected performance level. The third rule is not exactly a rule but a primary selection of Figures of Merit according to which the candidate performances will be compared as part of the Selection test results analysis. These multiple criteria are intended to provide a good picture of the relative performances of the proposed solutions.

Each rule is further described in the following sections:

PSS/MMS HBRAC Selection Rule 1:

Any candidate not compliant with all Design Constraints defined in PSS/MMS Audio Codec Design Constraints permanent document [1] will be excluded.

~~Same as PSS/MMS Low-Bit Rate Audio Codec Selection Rule 1 in Section 1.~~

PSS/MMS HBRAC Selection Rule 2:

Any candidate not meeting the performance requirements defined in [1] will be excluded.

~~Same as PSS/MMS Low-Bit Rate Audio Codec Selection Rule 2 in Section 1.~~

PSS/MMS HBRAC Selection Rule 3 Figure of Merits:

A number of Figures of Merit will be used to analyse and compare the performance of the candidates. Corresponding rankings will be prepared and provided for information only. None of the Figures of Merit listed below is intended to serve as single selection criteria.~~Same Figure of Merits as PSS/MMS Low-Bit Rate Audio Codec Selection Rule 3 in Section 1 except calculated for high bit rate test sets and high-rate content classes.~~

Preferred quality FoM:

For each test case *K* a delta performance score is calculated as the difference between the codec performance $\overline{P}_C(K)$ and the performance of a reference codec $\overline{P}_R(K)$:

$$\Delta_C(K) = \overline{P}_C(K) - \overline{P}_R(K)$$

The quality reference is calculated according to:

$$\overline{P}_R(K) = \tfrac{1}{N}\sum_{i\in I} P_R(K,i).$$

where *I* denotes the set of *N* test items.

The candidate codec performance is calculated according to:

$$\overline{P}_C(K) = \tfrac{1}{N}\sum_{i\in I} P_C(K,i).$$

The delta performance scores are arranged in a vector according to the test cases. Negative delta scores will be highlighted in red in order to indicate cases where the reference performance it is not met.

In addition to the item-wise delta scores per test cases, average, minimum and maximum delta scores will be given across test cases.

The minimum and maximum delta score is understood as worst, respectively, best observed score across all items in test cases in which the reference is AAC-LC.

In order to provide a global overview further composite scores are derived such as average, minimum and maximum scores across the complete set of test cases.

An overview of the complete matrix of scores is given in table 2:

Content type Operating condition	Audio	Min per item	Max per item
24 kbps, mono, use case A (PSS)			
24 kbps, stereo, use case A (PSS)			
32 kbit/s, stereo, use case A (PSS)			
32 kbps, stereo, use case B			
48 kbps, stereo, use case A (PSS)			
48 kbps, stereo, use case B			
32 kbps, stereo, use case A (PSS), 1% FER			
32 kbps, stereo, use case A (PSS), 3% FER			
Average			
Min per item			Not used
Max per item		Not used	

Informative quality FoM:

For informative quality FoM each test case *K* a delta performance score is calculated as the difference between the codec performance $\overline{P}_C(K)$ and the performance of a informative reference codec $\overline{P}_{IR}(K)$:

$$\Delta_C(K) = \overline{P}_C(K) - \overline{P}_{IR}(K)$$

The quality informative reference is calculated according to:

$$\overline{P}_{IR}(K) = \tfrac{1}{N}\sum_{i\in I} P_{IR}(K,i),$$

where *I* denotes the set of *N* test items.

The candidate codec performance is calculated according to:

$$\overline{P}_C(K) = \tfrac{1}{N}\sum_{i\in I} P_C(K,i).$$

The delta performance scores are arranged in a vector according to the test cases. Negative delta scores will be highlighted in red in order to indicate cases where the reference performance it is not met.

In addition to the item-wise delta scores per test cases, average, minimum and maximum delta scores will be given across test cases.

The minimum and maximum delta score is understood as worst, respectively, best observed score across all items in test cases in which the informative reference is RealAudio @ 32 and 48 kbit/s stereo.

In order to provide a global overview further composite scores are derived such as average, minimum and maximum scores across the complete set of test cases.

An overview of the complete matrix of scores is given in table 3:

Content type Operating condition	Audio	Min per item	Max per item
32 kbit/s, stereo, use case A (PSS)			
32 kbps, stereo, use case B			
48 kbps, stereo, use case A (PSS)			
48 kbps, stereo, use case B			
Average			
Min per item			Not used
Max per item		Not used	

FoM H1:

The number of positive delta MUSHRA table entries.

FoM H2:

The number of negative delta MUSHRA table entries.

FoM H3a-H3e:

Figure of Merit for computational complexity and memory are:

- The peak-WMOPS (measured for the worst observed frame)
- Average-WMOPS (measured over the test material)
- RAM (in kWords measured for the worst test case)
- ROM (in kWords measured for the worst test case)
- Program ROM (number of instructions measured for the worst test case)

3 PSS/MMS Audio Codec Selection Procedure

The selection procedure will consist of the following steps:

Low Bit-Rate codec discussion (steps 1-5):

1. The LBR Selection test results will be presented and analysed while keeping secret the identity of the LBR candidates. Each candidate will be informed of the code used for its own solution and its solution only. The Selection rules 2 and 3 defined in section 1 will be applied at this stage.
2. After the review and discussion of the test results (as specified for rule 3), TSG-SA4 will try to reach a consensus on a quality ranking of the LBR candidates.
3. Each LBR candidate will then present its solution and show the compliance with the PSS/MMS Audio Codec Design Constraints [1]. All candidates not compliant with all design constraints will be excluded according to the Selection rule 1.
4. The test results obtained by each LBR candidate will then be revealed.
5. A discussion and review of the LBR candidate codec characteristics and test results will take place.

High Bit-Rate codec discussion (steps 6-10):

6. The HBR Selection test results will be presented and analysed while keeping secret the identity of the HBR candidates. Each candidate will be informed of the code used for its own solution and its solution only. The Selection rules 2 and 3 defined in section 2 will be applied at this stage.
7. After the review and discussion of the test results (as specified for rule 3), TSG-SA4 will try to reach a consensus on a quality ranking of the HBR candidates.
8. Each HBR candidate will then present its solution and show the compliance with the PSS/MMS Audio Codec Design Constraints [1]. All candidates not compliant with all design constraints will be excluded according to the Selection rule 1.
9. The test results obtained by each HBR candidate will then be revealed.
10. A discussion and review of the HBR candidate codec characteristics and test results will take place.

Selection of PSS/MMS Audio codec(s) for low and high bit-rate ranges:

11. SA4 will try to reach a consensus on codec(s) for the PSS/MMS default audio codec for low and high bit-rate range.

In addition to the above selection procedure, all candidates have to provide the Selection Deliverables as defined in Annex A. All LBR and HBR candidates not compliant with the required deliverables will be excluded (before Step 1).

References:

[1] S4-030433 “PSS/MMS Audio Codec Selection, Design Constraints and Performance Requirements – Version 2.0”

[2] AMR-WB+ permanent document; Design Constraints (Last version approved by TSG-SA4)

[3] AMR-WB+ permanent document; Performance Requirement (Last version approved by TSG-SA4)

[4] PSS/MMS Audio Codec and AMR-WB+ permanent document; Time Plan (Last version approved by TSG-SA4)

[5] PSS/MMS Audio Codec and AMR-WB+ permanent document; AMR-WB+ and PSS/MSS low-rate audio selection test and processing plan (Last version approved by TSG-SA4)

Annex A: Selection Deliverables for PSS/MMS Audio Codec and Extended AMR-WB

1. Introduction

This Annex lists the deliverables for the selection phase for PSS/MMS Audio Codec and Extended AMR-WB. The deliverables are all items the candidates must provide in order to enter into the selection contest.

The delivery dates for all selection deliverables are based on schedule assumptions given in the permanent document on codec selection and development [4]. In case of any discrepancy of the dates, the dates as indicated in [4] prevail.

2. List of Deliverables

The candidates participating to the selection phase must provide the following deliverables:

- Binding declaration to submit a candidate codec
- Codec executable(s)
- Technical descriptions (including draft Specifications - to be distributed only by the winning proponent(s))
- Report covering the design constraints
- Source C-code for tested codec modes
- IPR declaration
- Optional additional information

Each item is described in the following sections.

In addition, for the verification phase (after the selection phase), the winning proponent(s) must submit he ANSI-C source code of selected codec(s) to verification labs (under NDA) .

2.1 Binding declaration to submit a candidate codec

The candidates must make the binding declaration (commitment to funding the selection phase) by 31st May 2003.

~~*[Editor's note: There was still some disagreement in audio codec ad-hoc#2 (June 2003) on sharing the cost of testing among the candidates. There may be something to clarify on the shares of the codec proponents.]*~~

2.2 Executable

The candidates must deliver to ETSI copies of their executable by October 17 2003~~[tbd]~~. It is the responsibility of the candidates to be sure that the executable will effectively be delivered by the due date. ETSI will register the executable delivery date for each candidate and will report the effective delivery date to SA4. ETSI will not check the correct operation of the files delivered.

The executables will be used by the host laboratories to create the samples used in the listening tests. ~~after the end of the selection to verify that the selected candidate solution is able to regenerate the processed samples used during the selection tests.~~

~~*[Editor's note: It is still to be decided how the processing will be carried out. If processing of samples is done by the proponents themselves in a shared manner, then further deadlines for processed samples and cross-checking of processing are needed.]*~~

2.3 Technical descriptions

The candidates must provide by February 18 2004~~[tbd]~~ a technical description of their codec through SA4 reflector. The description should contain sufficient details to allow analysis of the solution.

Each candidate shall also provide a report through SA4 reflector by February 18 2004~~[tbd]~~ showing that the proposal fulfils all design constraints. This includes a complexity evaluation based on the floating-point code: Worst Observed Frame for the codec, memory (~~scratchpad RAM, static~~ RAM and data ROM) and Program ROM estimates based on the floating-point implementation. The Worst Observed Frame figure must be computed from the complete database of material used for the selection phase.

In addition, each proponent shall have developed a draft version of the specification, but this is not a required deliverable before selection. Immediately following the selection at the SA4 meeting, the selected candidate(s) must publish this detailed description by providing a soft copy of the document to the SA4 secretary, who will make it available to meeting delegates and upload it onto the ETSI and 3GPP FTP sites. All SA4 organizations are then invited to comment and review the draft specification at the SA4 selection meeting (and in possible subsequent Audio Codec ad-hoc meeting ~~t.b.d~~).

The format of the specification is as described below:

- Detailed technical description of the encoder and decoder

- ANSI-C source code of the tested encoder and decoder

Source code for both encoder and decoder (in the form as tested) shall be under the copyright and control of 3GPP (or one of its organizational partners) and shall be available from the controlling entity (3GPP or one of its organizational partners) to interested companies having a bona fide interest in receiving such source code.

Diligent efforts will be made to establish an infrastructure of confidential disclosure of the software, in order to ensure that valuable implementation details and trade secrets are not used by non-eligible entities, not used for products not compliant with the 3GPP specification and not used for non-3GPP applications. Access to the software may be subject to a reasonable fee paid to the controlling entity and will be granted solely for applications within the scope of the respective 3GPP specification. (Wording to be reviewed by 3GPP legal adviser.)

~~*[to be added]*~~

2.4 Source C-code (for the tested codec mode)

The candidates must deliver to ETSI a disk containing a copy of their ANSI C-Code used for the processing of the samples so that it arrives at ETSI by February 27 2004~~[tbd]~~.

The compiled versions of the source C-Code and the executables delivered to ETSI (see Section 2.2) should give identical and bit-exact versions of all samples used for the selection phase. This version of the code should allow a third party to re-process the samples in order to check the integrity of the material used for the selection tests.

This C-code will be used to check the complexity estimates of the proposal. To that purpose, the candidate must also provide the following information for the solution:

1) Data RAM

- For each source file, enumeration of static variables, types and their associated length;
- Function call path leading to largest scratch RAM usage and list of temporary variables active in that case

2) Data ROM

- for each source file, enumeration of tables, types and their associated length

3) Program ROM

- list of source files (.c, .h)
- number of pure instruction C lines for each .c file

4) wMOPS

- The C source code should contain instrumentation and counters for basic operations, data move, logical operations and arithmetic tests.

- Sample and experiment condition that produced the highest wMOPS figure

2.5 IPR Declaration~~?~~

The candidates must provide by February 18 2004~~[tbd]~~ a mutually acceptable declaration of IPR. Formal IPR declaration shall be submitted.

Candidates are advised to discuss the form of this IPR statement with the corresponding 3GPP Organisational Partner ~~ETSI Legal Adviser (see below)~~ well in advance of this date, to define what is mutually acceptable, e.g. ETSI Legal Adviser (see below).~~.~~

Mr. Stephane Tronchon
ETSI Legal Adviser
ETSI / PT SMG
650 Route des Lucioles
06921 Sophia Antipolis Cedex
France
Email: stephane.tronchon@etsi.org~~fr~~

~~The written statement can also be sent by Fax before the deadline.~~

A copy of the statement must be sent to Mr. Paolo Usai at the following address: paolo.usai@etsi.org~~fr~~

2.6 Optional additional information

The candidates are free to provide any additional information likely to help in the evaluation of their proposal by February 18 2004.

References

See reference list in the main body of this document.