TSG-SA WG4 (SA4) - CODEC Status Report at TSG-SA#24

Kari Järvinen
TSG-SA WG4 Chairman

- ☞ *SA4 status report in Tdoc SP-040341*
- ☞ *Slides in Tdoc SP-040409*

A GLOBAL INITIATIVE

Content

- General issues ☜
- Review of SA4 work progress for Release 6
- Maintenance of releases
- Communication with other groups
- Documents for information
- Documents for approval

General: SA4 officials

• **Chairman:**	**Kari Järvinen (Nokia, ETSI)**
• **Vice Chairpersons:**	**Catherine Quinquis (Orange, ETSI) and Frédéric Gabin (NEC Technologies, ETSI)**
• **Secretary:**	**Paolo Usai (3GPP Support)**

- **Sub Working Groups / Ad-Hoc groups:**
  - **Speech Quality (SQ) SWG** — **Paolo Usai (ETSI)**
  - **PS Multimedia (PSM) SWG** — **(open)**
  - **Audio Codec Ad-Hoc group** — **Imre Varga (Siemens, ETSI)**
  - **Video Codec Ad-Hoc group** — **Nikolaus Färber (Fraunhofer Gesellschaft, ETSI)**

Kari Järvinen (Nokia, ETSI) was re-elected as SA4 Chairman at SA4#31 (in May) for another two-year term by acclamation.

The PSM SWG Chairman Rolf Hakenberg (Panasonic, ETSI) stepped down after SA4#31. Thanks to Rolf on behalf of SA4 for the excellent work as PSM SWG Chairman!

General: SA4 meetings

- **Meetings held**
  - **PSM SWG ad-hoc #5** — **5 - 6 April, 2004** — **Host: Ericsson; Venue: Lund, Sweden**
  - **Video codec ad-hoc #2** — **7 April, 2004** — **Host: Ericsson; Venue: Lund, Sweden**
  - **SA4#31** — **17 - 21 May, 2004** — **Host: VoiceAge; Venue: Montreal, Canada**
  - **SA4 audio codec ad-hoc** — **4 June, 2004** — **Host: Coding Technogies; Venue: Nuremberg, Germany**

- **Future meetings**
  - **SA4#32** — **16 - 20 August, 2004** — **Host: The European Friends of 3GPP; Venue: Prague, The Czech Republic**
  - **SA4 ad-hoc on MBMS with SA3 delegates invited** — **23 August (tbc), 2004** — **Host: tbd; Venue: tbd**
  - **SA4#33** — **22 - 26 November, 2004** — **Host: The European Friends of 3GPP; Venue: Helsinki, Finland**

- **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
SA4#29	167	53	18	8
SA4#30	215	74	27	9
SA4#31	**168**	**57**	**26**	**7**

General: Input documents

- **For information:**
  - **SP-040341: TSG S4 Status Report at TSG-SA#24; Source: SA4 Chairman**
  - **SP-040409: Slides presentation of the SA4 status; Source: SA4 Chairman**
  - **SP-040425 - SP-040433: New audio codec draft TSs v1.0.0 (Release 6); Source: SA4**
- **For approval:**
  - **SP-040342: 3GPP TR 26.935 "Packet Switched Conversational Multimedia Applications; Performance Characterisation of Default Codecs" Version 2.0.0 (Release 6); Source: SA4**
  - **SP-040343: 3GPP TS 26.243 "ANSI-C code for the Fixed-Point Distributed Speech Recognition Extended Advanced Front-end" Version 2.0.0 (Release 6); Source: SA4**
  - **SP-040344: 3GPP TS 26.245 "Transparent end-to-end packet switched streaming service (PSS); Timed text format" Version 2.0.0 (Release 6); Source: SA4**
  - **SP-040345: 3GPP TS 26.246 "Transparent end-to-end packet switched streaming service (PSS); 3GPP SMIL Language Profile" Version 2.0.0 (Release 6); Source: SA4**
  - **SP-040434: CRs TS 26.234 on “addition of Release 6 functionality” (Release 6); Source: SA4**
  - **SP-040356: CRs TS 26.235 and TS 26.236 on “the introduction of the DSR codec” (Release 6); Source: SA4**
  - **SP-040357: CRs TS 26.236 on "RTCP usage for IMS" (Release 5 and Release 6); Source: SA4**

Content

- **General issues**
- **Review of SA4 work progress for Release 6**
- **Maintenance of releases**
- **Communication with other groups**
- **Documents for information**
- **Documents for approval**

Release 6 WIs

- Performance Characterisation of Default Codecs for PS Conversational Multimedia Applications
- Packet Switched Streaming (PSS) Rel-6
- MMS Enhancements: MMS codecs and formats
- 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
- MBMS User Services: Definition of MBMS user services, media codecs, formats and transport/application protocols using Multimedia Broadcast/Multicast Service (MBMS)
- Codec Enhancements for Packet Switched Conversational Multimedia Applications
- 3G-324M Improvements (CS Multimedia Telephony Service Terminal)

Performance Characterisation of Default Codecs for PS Conversational Multimedia Applications

- **TR 26.935 "PS conversational multimedia applications; Default codecs; Performance characterization" for approval in Tdoc SP-040342.**
- **Gives information of the performance of the default speech codecs (AMR-NB and AMR-WB) in PS conversational applications under various operating and transmission conditions. (Also several ITU-T speech codecs were included in Phase 2.)**
- **Testing from October 2003 until February 2004. The results reported and the testing work approved at SA#22 (for Phase 1) and at SA#23 (for Phase 2).**
- **Results confirm that the default speech codecs (AMR-NB and AMR-WB) operate well for PS conversational applications over various realistic operating conditions (i.e., packet loss, delay, background noise, radio conditions and use of ROHC).**
- **The results also indicate that users have clear preference for AMR-WB speech over AMR-NB speech.**
- **The preparation of the TR completes the work under this WI.**

Packet Switched Streaming (PSS) Rel-6

- **TS 26.245: "Transparent end-to-end packet PS service (PSS); Timed text format" for approval in Tdoc SP-040344**
  - Timed text is text that is rendered at the terminal, in synchronization with other timed media such as video or audio. Downloaded, not streamed.
  - Rel-6 contains (optional) text wrap and some minor updates to Rel-5.

- **TS 26.246: "PSS; 3GPP SMIL Language Profile" for approval in Tdoc SP-040345**
  - SMIL is used for the description of the spatial layout and temporal behaviour of a presentation.
  - Compared to SMIL in Rel-5, the set of high-level features remains unchanged. The main additions for Rel-6 are in details.

- **CR on 'addition of Release 6 functionality' to TS 26.234 "PSS; Protocols and Codecs" for approval in Tdoc SP-040434.**
  - Brings new Rel-6 features and completes restructuring of this TS (some functionalities moved into their own Rel-6 TSs). The new functionalities include:
    - **support for media stream selection by clients**,e.g., to allow sessions with several bit-rates or languages to be offered by the server
    - **support for bitrate adaptation**; buffer feedback from client is used by the server to achieve enhanced robustness for varying transmisson rates and enables bit-rate switching
    - **extended support for synthetic audio**; SP-MIDI Mobile Downloadable Sounds
    - **support for Quality of Experience (QoE) metrics**; for servers to receive information from the handset to provide the service providers means to evaluate the end user experience
    - **clarifications, updates and extensions to a number of protocols** (SDP, RTSP, RTP, RTCP)
    - **update of PSS UAProf vocabulary and RDF schema** to identify the PSS base applications and to express detailed client capabilities for all release-6 functionalities
    - **support for progressive downloading** (playing while downloading)

Packet Switched Streaming (PSS) Rel-6: Video codecs

- **Except for one company, SA4 has agreed ITU-T H.264 (MPEG-4 AVC) as the working assumption for recommended ("should be supported") video codec for adoption to PSS - and also to other services: MMS, PS conversational applications and 3G-324M (CS multimedia terminal).**
- **Draft specification text to adopt H.264 (AVC) for PSS (into TS 26.234), for MMS (TS 26.140) and for PC conversational applications (TS 26.235) and also on the impact to File Format (TS 26.244) prepared and found agreeable for all in SA4 except for one company.**
- **Some technical details remain to be solved. Further discussions needed to resolve these at next SA4 meeting to complete the video codec definition for Rel-6.**
- **Some companies asked for further testing. The rest of SA4 sees that sufficient evidence has been already presented in SA4 and no additional evidence is needed to adopt H.264 (AVC) as a recommended video codec for PSS (and for the other above mentioned services).**

Packet Switched Streaming (PSS) Rel-6: Audio codecs

- **Two recommended PSS audio codecs found acceptable at SA#23, after some discussion.**
  - Two codec candidates (Enhanced aacPlus and Extended AMR-WB) met the design constraints and quality performance requiments for PSS and MMS.
  - No consistent ranking was possible. (The PSS/MMS audio codec codec selection test results and global analysis were reported to SA#23 in Tdocs SP-040073 and SP-040072.)
  - SA4 aim to recommend two codecs (“should be supported”) instead of defining default codec(s) (“shall be supported” i.e. mandatory support) was explained; draft specification text for PSS TS 26.234 presented for information.
  - Codec selection was not yet brought for formal approval since the critical verification tasks for two codecs could not be completed in time for SA#23.
- **SA4#31 agreed on the selection of these two recommended audio codecs for PSS.**
  - Upon agreeing on the codecs in SA4, draft versions of the new codec TSs were given for review to SA4.
  - Critical verification work (e.g., complexity, bit-exactness) based on C-codes launched; to be carried out by SA#24. Source C-codes distributed under NDA to verification organisations.
  - Presentation of codec selection (CR to TS 26.234 to define the new codecs for PSS) and new codec specifications for approval targeted already at SA#24.

Packet Switched Streaming (PSS) Rel-6: Audio codecs

- **SA4 ad-hoc meeting on audio codecs was scheduled for June 4th (still before SA#24):**
  - SA4#31 gave authority to approve the verification work results and the new audio codec TSs on behalf of SA4.
  - Presentation of codec selection (CR to TS 26.234) and new codec specifications for SA#24 approval pending on approval of the codecs to pass critical verification and agreement on the new TSs at the ad-hoc meeting.

- **The critical verification items (before codecs brought for SA approval):**
  1. **Verification of bit-exactness of C-code** (to be put into specifications) to the executable used for selection testing and to the executable used for freezing the codec (delivered to ETSI before testing started); over the selection test audio material. This ensures that 1) the source C-code is of the "winning codec" and it produces the exact quality (bit-exact audio samples) demonstrated in selection tests and 2) the codec was not tuned for any particular test samples during testing. Carried out by Siemens in collaboration with the ETSI MCC SA4 Secretary.
  2. **Output sampling rate at 8 kHz** - the decoder to be able to produce an output signal at 8 kHz, irrespective of the input signal sampling frequency. Carried out by Siemens.
  3. **Complexity check against design constraints**: Candidates cross-check, STMicroelectronics
  4. **Verification of the format of the C-code**: Candidates cross-check, STMicroelectronics
  5. **Verification of error insertion device and error concealment** used for testing: Candidates cross-check STMicroelectronics
  6. **Review of draft TSs**: by all organizations in SA4 (draft TSs were distributed over SA4 reflector)

Packet Switched Streaming (PSS) Rel-6: Audio codecs

- **Verification results (SA4 audio codec ad-hoc meeting on 4th June):**
  - Item 1: Extended AMR-WB passed, but Enhanced aacPlus failed (in 3 out of 188 samples).
    - Coding Technologies proposed a solution to resolve the problems of bit-exactness by submitting a new verification executable. However, due to time constraints, procedural concerns (checksums), and the mandate of the ad-hoc, this was not accepted by Nokia and Ericsson.
  - Items 2-5: Both codecs passed
    - Some minor problems were reported for the Enhanced aacPlus codec for item 2.
  - Item 6: For both codecs, the TSs are not yet mature enough for approval; draft TSs to be presented only for information to SA#24. Some specific concerns were expressed:
    - Ericsson and Nokia felt unable to approve the enhanced aacPlus specifications because the decoder is specified by reference to MPEG only, which would prevent 3GPP from applying CRs to the specification and which would make the decoder specification of enhanced aacPlus unclear to whether it can decode low bit rate AAC+ bitstreams. It was also felt by Ericsson that a codec that failed a bit exactness test is not eligible for specification approval. Clarification was requested.
    - Coding Technologies felt unable to approve the AMR-WB+ codec specification because it was unclear whether the specified codec would operate inline with the AMR-WB+ design constraints and whether it would comply with the complexity figures used in the selection process.

- **Draft audio codec TSs presented "only" for information at SA#24:**
  - In Tdocs SP-040425 - SP-040427 for Extended AMR-WB
  - In Tdocs SP-040428 - SP-040433 for Enhanced aacPlus

Packet Switched Streaming (PSS) Rel-6: Audio codecs

- **Further audio codec work**
  - Issues in critical verification to be solved: Target to bring the codec TSs for approval at SA#25 with a corresponding CR to TS 26.234 (to define the new codecs for PSS).
  - Verification ("non-critical") and characterisation testing (e.g., checking of the C-code for any remaining bugs, complete detailed complexity analysis, calculation of frequency response, performance for special input voices and background noises).
  - Fixed-point implementations of the codecs to be produced. Both proponents committed to provide fixed-point implementation offering audio quality not significantly different from the existing floating-point implementation and both also expect the fixed-point implementation to meet the design constraints set for the floating-point implementation
  - Compliance requirements to specifications
  - Prepare TR on audio codecs performance characterisation to give information of the performance of the two recommended codecs (based on selection tests, verification tests and further complementing characterisation tests).

MMS Enhancements: MMS media formats and codecs

- **During SA4#31, MMS audio codec selection debated extensively**
  - Most companies (in SA4 audio codec ad-hoc group session) stated preference for choosing one default ("shall be supported" i.e. mandatory support)
    - This was seen bringing the benefit of reducing implementation costs. Some companies stated it also guaranteed interoperability. Some companies however pointed out that interoperability to terminals of earlier releases is not guaranteed.
  - Support given also for other options, e.g., for two recommended codecs, for one default encoder and two default decoders.

- **Both candidate codecs (Enhanced aacPlus and Extended AMR-WB) have merits over the other depending on the bit-rate and content type like explained to SA#23**
  - Both met PSS/MMS design constraints and requirements for performance (audio quality).
  - Choosing between them is difficult and a matter of preferences between bit-rates and content types (use cases). No consensus reached at SA4#31 on which codec to choose.
  - Proposal for defining two recommended codecs for MMS (like for PSS) put for approval by correspondence after SA4#31 (by Wednesday 26th May) - not agreed due to 3 objections (T-Mobile, Telecom Italia and Orange)

- **No agreement on single MMS codec was reached at SA4#31. Guidance from SA and relevant WGs (T2, SA1 on use cases) likely needed.**

- **On MMS video codecs, except for one company, SA4 has agreed H.264 (AVC) as working assumption to be adopted as recommended ("should be supported") codec.**

Extended AMR-WB codec (AMR-WB+) Targeted for PS Streaming and Messaging Services

- **AMR-WB+ codec developed under this work item is considered as candidate audio codec for PSS and MMS.**
- **Work related to ongoing SA4 audio codec selection work for PSS and MMS as the AMR-WB+ codec is considered as one candidate for PSS and MMS audio codec and the testing of all codec candidates has been carried out as combined testing.**
- **AMR-WB+ codec specifications are presented for information to SA#24.**

Codec Work to Support Speech Recognition Framework for Automated Voice Services (SES codecs)

- **Verification tests completed: both candidate codecs met the requirements**
  - bit-exactness of the codec for specifications against the one used during selection testing
  - implementation complexity meeting design constraints
- **Therefore, SA4 asks SA approval for the SES codec selection:**
  - DSR Extended Advanced Front-end ("should be supported") and AMR or AMR-WB ("may be supported"); with substantial performance advantage from DSR noted
  - CRs to TSs 26.235 and 26.236 (defining the use of SES codecs) in Tdoc SP-040356
  - new SES codec specification TS 26.243 ("ANSI-C code for the Fixed-Point DSR Extended Advanced Front-end") in Tdoc SP-040343.
- **Draft version of the new TS and of the intended formulation of the two CRs presented for information at SA#23. Some editorial improvements done since then.**
- **TR on SES codec characterisation planned based on the test results obtained during the selection and verification phases. To be then referenced in TSs 26.235.**
- **The SA4 work is completed apart from the "non-critical" TR.**

Media Codecs and Formats for IMS Messaging and Presence

- **A “skeleton” working draft of TS 26.141 “IMS Messaging and Presence; Media formats and codecs” was prepared at SA4#28 (September 2003).**
- **Since then there has been no contributions or progress for this WI.**

MBMS User Services

- **Application level Forward Error Correction (FEC) and proposed solutions discussed further.**
  - Simulation guidelines under preparation with relevant RAN and GERAN WG; to be used to evaluate and compare the FEC proposals. Target to make selection at next SA4 meeting.
  - SA4 sees that FEC should be supported in an MBMS service infrastructure. Requirements for FEC support for the terminal to be discussed at next SA4 meeting.
- **For reliable transmission (in downloading), SA4 sees point-to-point (ptp) repair is needed in addition to FEC. (Point-to-multipoint (ptm) repair is for further study.)**
- **Protocol definitions progressed**
  - Clarifications made on the use of dowload delivery methods (e.g. requirements set for support for the different features in FLUTE) and on the use of SDP.
- **Some discussion on MBMS security with SA3**
  - On the request of SA3, comments given on the suitability of methods for protecting MBMS streaming (e.g. IETF SRTP) and downloading.
  - SA4 ad-hoc meeting on MBMS planned on 23rd August (to be confirmed); SA3 delegates have been invited to participate.
- **On MBMS codecs, initial proposals have been made for adopting H.263 or H.264 (AVC) as a single default video codec, and adopting new audio codec.**
  - Harmonisation with other services is felt important, and the developments and performance analysis in new codec selections for other services (PSS, MMS) are being followed.
- **TS 26.346 “MBMS Protocols and Codecs” progressed. To be presented for information at SA#25 in September. Finalisation expected by December 2004.**

Codec Enhancements for Packet Switched Conversational Multimedia Applications

- **This WI approved at SA#23 considers enhancements for the set of codecs (and the related transport protocols) for PS Conversational Multimedia Applications.**
- **Except for one company, SA4 has agreed H.264 (AVC) as working assumption to be adopted as recommended ("should be supported") video codec for PS conversational multimedia applications into TS 26.235.**

3G-324M Improvements

- **This WI approved at SA#23 considers a number of backwards-compatible updates to the 3G-324M (CS multimedia telephony service terminal).**
- **At SA4#31, addition of optional H.264 (AVC) and AMR-WB support proposed. Draft CR text to TS 26.111 "Codec for CS Multimedia Telephony Service; Modifications to H.324" formulated on optional H.264 and AMR-WB.**
- **No formal agreements yet taken and comments invited until SA4#32 for finalisation of the CR.**

Content

- **General issues**
- **Review of SA4 work progress for Release 6**
- **Maintenance of releases**
- **Communication with other groups**
- **Documents for information**
- **Documents for approval**

Maintenance of releases

- **A CR to TS 26.236 (Rel-5 and Rel-6) on "RTCP usage for IMS" is brought for approval in Tdoc SP-040357.**
  - Optimisation of Voice over IMS discussed due to request from SA2 to study and comment on how to efficiently handle RTCP associated to an RTP flow (to obtain bandwidth savings and avoid disruptions).
  - RTCP is required for instance to synchronise multiple media streams, and in multi-party RTP sessions. However, for a point-to-point speech only service, SA4 sees that RTCP is not always required.
  - SA4 recommends that RTCP packets should be sent for all types of multimedia sessions except for point-to-point speech only sessions. For point-to-point speech only sessions, a UE should not send RTCP packets. This avoids RTCP disrupting the RTP speech flow and causing impairment to speech quality.
  - This application level solution solves Voice over IMS RTCP usage problems, but only for point-to-point speech only service.
    - SA4 has informed SA2, RAN2 and RAN3 of the above SA4 view (in LS sent in mid-April, approved by correspondence in SA4) and asked these groups to continue to work on solutions for efficient transport of RTCP.

Content

- **General issues**
- **Review of SA4 work progress for Release 6**
- **Maintenance of releases**
- **Communication with other groups**
- **Documents for information**
- **Documents for approval**

Communication with other WGs/TSGs/groups

Tdoc no.	Title	Intended for	Copy to
TD S4-040203*	Reply LS to LS on Optimisation of Voice over IMS	TSG SA WG2, TSG RAN WG2	RAN WG3
TD S4-040320	Reply LS on “Answer to MBMS ARP Support in UTRAN”	TSG RAN WG2, TSG RAN WG3, TSG SA WG2	TSG GERAN, TSG CN WG1
TD S4-040324	Reply LS on AMR mode selection for MMS	OMA Messaging Working Group (MWG) Multimedia Messaging Subgroup (MMSG)	3GPP2 TSG-C
TD S4-040355	Reply LS on MBMS support in UTRAN	TSG RAN WG2, TSG SA WG2	TSG GERAN, TSG CN WG1, TSG RAN WG3
TD S4-040322	Reply LS on MBMS security issues	TSG SA WG3	TSG SA WG2, OMA DL+DRM
TD S4-040356	Response LS on Multiple MBMS Issues	TSG RAN WG1, TSG RAN WG2, TSG RAN WG3, TSG GERAN, TSG GERAN WG2	
TD S4-040347	LS on Optimisation of Voice over IMS	TSG CN WG1, TSG RAN WG2	
TD S4-040309	Liaison statement on DRM protection for PSS	TSG SA WG3	OMA-BAC DL+DRM, ISMA

) drafted in PSM ad-hoc meeting in April and approved by correspondence in SA4 (before SA4#31).

Content

- General issues
- Review of SA4 work progress for Release 6
- Maintenance of releases
- Communication with other groups
- Documents for information
- Documents for approval

Documents for information

- **Extended AMR-WB codec specifications**
  - Tdoc SP-040425: 3GPP TS 26.273 "Fixed-point ANSI-C code for the Extended Adaptive Multi-Rate - Wideband (AMR-WB+) codec" Version 1.0.0 (Release 6)
  - Tdoc SP-040426: 3GPP TS 26.290 "Audio codec processing functions; Extended Adaptive Multi-Rate - Wideband (AMR-WB+) codec; Transcoding functions" Version 1.0.0 (Release 6)
  - Tdoc SP-040427: 3GPP TS 26.304 "Floating-point ANSI-C code for the Extended Adaptive Multi-Rate - Wideband (AMR-WB+) codec" Version 1.0.0 (Release 6)

- **Enhanced aacPlus codec specifications**
  - Tdoc SP-040428: 3GPP TS 26.401 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; General description" Version 1.0.0 (Release 6)
  - Tdoc SP-040429: 3GPP TS 26.402 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Additional decoder tools" Version 1.0.0 (Release 6)
  - Tdoc SP-040430: 3GPP TS 26.403 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Encoder specification; Advanced Audio Coding (AAC) part" Version 1.0.0 (Release 6)
  - Tdoc SP-040431: 3GPP TS 26.404 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Encoder specification; Spectral Band Replication (SBR) part" Version 1.0.0 (Release 6)
  - Tdoc SP-040432: 3GPP TS 26.405 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; Encoder specification; Parametric stereo part" Version 1.0.0 (Release 6)
  - Tdoc SP-040433: 3GPP TS 26.410 "General audio codec audio processing functions; Enhanced aacPlus general audio codec; ANSI-C code" Version 1.0.0 (Release 6)

Content

- General issues
- Review of SA4 work progress for Release 6
- Maintenance of releases
- Communication with other groups
- Documents for information
- Documents for approval

Documents for approval

- **Tdoc SP-040342: new TR 26.935 "Packet Switched Conversational Multimedia Applications; Performance Characterisation of Default Codecs" Version 2.0.0 (Release 6)**
  - Gives information of the performance of the default speech codecs (AMR and AMR-WB) in PS conversational multimedia applications under various operating and transmission conditions.
  - A draft version 1.0.0 of the TR was presented for information at SA#23. A number of updates/corrections made.
  - Testing was divided into 2 phases. Phase 1 considered the default speech codecs AMR-NB and AMR-WB in various operating conditions. Phase 2 considered also several ITU-T codecs.
  - The results confirm that the default speech codecs (AMR-NB and AMR-WB) operate well for packet switched conversational multimedia applications over various realistic operating conditions (i.e., packet loss, delay, background noise, radio conditions and with ROHC).
  - The quality is somewhat reduced when packet losses occur and the end-to-end delay is increased, but the overall quality still remains acceptable even with 3% packet loss rate in the terrestrial IP network and up to a maximum of 1% BLER on each radio leg.
  - The results also indicate that users have clear preference for AMR-WB speech over AMR-NB speech.
  - The performance results can be used as guidance for network planning regarding the QoS parameters for VoIP.

Documents for approval

- **Tdoc SP-040344: new TS 26.245 "Transparent end-to-end packet switched streaming service (PSS); Timed text format" Version 2.0.0 (Release 6)**
  - Timed text is text that is rendered at the terminal, in synchronization with other timed media such as video or audio.
  - The format of timed text is defined for downloaded files. (Timed text is downloaded, not streamed.)
  - Timed text format for Rel-6 contains (optional) text wrap and some minor updates from Rel-5.

- **Tdoc SP-040345: new TS 26.246 "Transparent end-to-end packet switched streaming service (PSS); 3GPP SMIL Language Profile" Version 2.0.0 (Release 6)**
  - Specifies the 3GPP SMIL (Synchronized Integrated Multimedia Language) Language Profile. Used for the description of the spatial layout and temporal behaviour of a presentation. A markup language based on SMIL 2.0 Basic and SMIL Scalability Framework (from W3C); subset of SMIL 2.0 Full profile and a superset of SMIL 2.0 Basic. Used by the PSS and MMS services, but not restricted to be used with only these services.
  - Compared to SMIL in Rel-5, the set of high-level features remains unchanged. The main additions for Rel-6 are in details such as:
    - media Parameters module from SMIL 2.0 and specification how to use it
    - more details on how to use systemCaption attribute of SMIL 2.0 BasicContentControl module
    - new URIs to check for 3GPP Rel 6 document conformance and SMIL player conformance

Documents for approval

- **Tdoc SP-040434: CRs TS 26.234 (“PSS; Protocols and Codecs”) on Addition of Release 6 functionality**

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.234	068	1	Rel-6	Addition of Release 6 functionality	B	5.6.0	S4	TSG-SA WG4#31	S4-040307

  - TS 26.234, existing in Rel-5, is updated through CRs.
  - A CR on “addition of Release 6 functionality” brings new Rel-6 features and completes restructuring of this TS.
  - New functionalities in Rel-6:
    - **support for media stream selection by clients** (alternatives in SDP); e.g., to allow sessions with several bit-rates or languages to be offered by the server
    - **support for bitrate adaptation**; buffer feedback from client is used by the server to achieve enhanced robustness for varying transmisson rates and enables bit-rate switching. On the client side, playout interruptions and overfillinging of buffers can be avoided or minimized.
    - **extended support for synthetic audio**; SP-MIDI has been complemented with Mobile Downloadable Sounds
    - **support for Quality of Experience (QoE) metrics**; for servers to receive information from the handset to provide the service providers means to evaluate the end user experience
    - **clarifications, updates and extensions to a number of protocols** (SDP, RTSP, RTP, RTCP)
    - **update of PSS UAProf vocabulary and RDF schema** to identify the PSS base applications (pure RTSP/RTP-based streaming, download, and SMIL presentation) and to express detailed client capabilities for all Release-6 functionalities
    - **support for progressive downloading** (playing while downloading)
  - The CR also completes restructuring of this TS (some functionalities moved into their own Rel-6 specifications): 1) 3GPP file format (3GP) to TS 26.244 approved at SA#23, 2) 3GPP Timed text format to TS 26.245 for approval at SA#24, and 3) 3GPP SMIL Language Profile to TS 26.246 for approval at SA#24.
  - Complementing CR(s) to TS 26.234 are still expected for SA#25, e.g., on Digital Rights Management extensions (RTP payload format for encryption, integrity protection of RTP).

Documents for approval

- **Tdoc SP-040343: new TS 26.243: "ANSI-C code for the Fixed-Point Distributed Speech Recognition Extended Advanced Front-end" Version 2.0.0 (Release 6)**

- **Tdoc SP-040356: CRs TS 26.235 and TS 26.236 ("PS Conversational Multimedia Applications; Default Codecs" and "PS Conversational Multimedia Applications; Transport Protocols")**

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.235	006	4	Rel-6	Introduction of the DSR codec	B	6.0.0	S4	TSG-SA WG4#31	S4-040360
26.236	010	3	Rel-6	Introduction of the DSR codec	B	5.4.0	S4	TSG-SA WG4#31	S4-040359

- **Tdoc SP-040357: CRs TS 26.236 on "RTCP usage for IMS" (Release 5 and Release 6)**

Spec	CR	Rev	Phase	Subject	Cat	Vers	WG	Meeting	S4 doc
26.236	011	1	Rel-5	RTCP usage for IMS	F	5.4.0	S4	TSG-SA WG4#31	S4-040345
26.236	012		Rel-6	RTCP usage for IMS	A	5.4.0	S4	TSG-SA WG4#31	S4-040346

  - RTCP is required e.g. to synchronise multiple media streams, and in multiparty RTP sessions. However, for a point-to-point speech only service, SA4 sees RTCP not always required.
  - SA4 recommends that RTCP packets should be sent for all types of multimedia sessions except for point-to-point speech only sessions. For point-to-point speech only sessions, a UE should not send RTCP packets. This avoids RTCP disrupting the RTP speech flow and causing impairment to speech quality.

(end of presentation)