Joint TSG-S4#9 - SMG11#14 Meeting *S4/SMG11 Tdoc (00)0035R2*
January 24-28 2000, Puerto Vallarta, Mexico

Liaison To:	**ITU-T SG16 Q20/16 “Audio & Wideband Coding”**
From:	**3GPP TSG-SA4 & ETSI SMG11**
cc:	**3GPP TSG SA, ITU-T SG 11**
Subject:	**Collaboration on Wideband Speech Codec development**

3GPP TSG-SA4 (Codec Working Group) and ETSI SMG11 would like to thank ITU-T SG16 Q20/16 for their recent communication proposing a closer collaboration between 3GPP and ITU-T on their on-going standardization of wideband speech codecs. TSG-S4 and SMG11 welcome the ITU-T Q20/16 proposal and appreciate the opportunity to share additional information on our respective projects with the objective to standardize harmonized solutions between ITU-T and 3GPP.

In the past few months, TSG-S4/SMG11 has focused its activity on the development of the Performance Requirements and Design Constraints for the wideband speech codec and established a detailed project plan for the codec selection and completion of the standard. In this process, TSG-S4/SMG11 recognized that potential codec proponents were equally interested in the 3GPP and ITU-T projects and that it was therefore critical to keep the requirements set by 3GPP and ITU-T as close as possible.

The detailed Performance Requirements and Design Constraints agreed in TSG-S4/SMG11 are provided in Attachments 1 and 2.

Although 3GPP believe that the requirements set for the wideband speech codec are equally challenging to those agreed by ITU-T in its wideband Audio Speech Codec Terms of Reference, we also identified a number of discrepancies exposed below.

First, the 3GPP wideband speech codec is intended to operate on existing or evolved 2G and 3G wireless networks. The compatibility with the existing GSM network architecture and design or with a standard handset platform induce a number of constraints, especially on the codec data rates and acceptable algorithmic complexity. These constraints are considered absolutely critical to the future commercial success of any codec used in a GSM Network. The following system applications were identified for this codec:

- A GSM full-rate traffic channel (22.8 kbit/s gross bit-rate) with an additional constraint of 16 kbit/s A-ter sub-multiplexing
- B GSM full-rate traffic channel (22.8 kbit/s gross bit-rate)
- C EDGE phase II channels
- D GSM multi-slot traffic channels (n*22.8 kbit/s)
- E 3G UTRAN channels

Unfortunately, some of these applications are not compatible with the ITU-T primary rates of 16 kbit/s and 24 kbit/s. TSG-S4/SMG11 has considered the possibility to introduce a similar requirement in our Wideband Speech Codec Design Constraints, but has found that this could result in additional complexity and ultimately to a sub-optimal solution for the targeted applications. Although this possibility has not totally been ruled out, 3GPP would rather not specify any mandatory rate. Specifically, the attached version of the Design Constraints document requests that the candidate proponents provide coders able to operate at or below the primary bit rates of interest to ITU-T Q20/16. And we would like to kindly ask ITU-T to consider the possibility to adopt a similar approach and accept that proponents provide solutions below the primary rates defined in the ITU-T Terms of Reference without changing the overall performances level of their solution.

Another point of concern is related to the development schedule and availability of the standard. The current working assumption in 3GPP is to complete the codec selection by 3Q00 and to finalize the core specifications before the end of the year. We would appreciate to receive additional information on the current ITU-T project time plan.

3GPP is looking forward to a continuous exchange of information with ITU-T SG16 Q20/16 on our respective wideband speech codec development projects. We believe that this will provide the basis for a closer collaboration, with the potential to lead us to an harmonized standard.

TSG-S4/SMG11 will next meet from February 28 to March 3, 2000, and is planning to review at that time the Design Constraints of the wideband codec taking into consideration any input to be received from ITU-T Q20/16.

Joint TSG-S4#9 - SMG11#14 Meeting
January 24-28, 2000, Puerto Vallarta, Mexico

S4/SMG11 Tdoc 90/00

Source:	**Editor**
Title:	**AMR wideband performance requirements (WB-3) version 1.2**
Document for:	**Review**
Agenda Item:	**7**

Introduction

This document contains the performance requirements for the AMR WB speech coder.

The performance requirements are defined for static and dynamic error conditions as well as speaker dependency, tandeming and input level dependency.

The requirements define the minimum acceptable performance of the candidate algorithm. Candidates are expected to pass all of the requirements. Objectives identify areas where particular emphasis should be placed by candidate developers who have met the requirements.

1. Definitions

The following systems/applications have been identified:

- A GSM full-rate traffic channel (22.8 kbit/s gross bit-rate) with an additional constraint of 16 kbit/s A-ter sub-multiplexing
- B GSM full-rate traffic channel (22.8 kbit/s gross bit-rate)
- C EDGE phase II channels
- D GSM multi-slot traffic channels (n*22.8 kbit/s)
- E 3G UTRAN channels

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

2. Requirements and Objectives for Applications A and B

2.1. Static conditions

Static conditions refer to channel cases where there is no shadowing. The speech quality of the codec modes applicable to the TCH-FS channel will be assessed over a range of C/I and background noise conditions to provide a ‘family’ of performance curves.

Requirements and objectives are specified for clean speech and background noise. The requirements and objectives for the TCH-FS traffic channels under static test conditions are specified in Table 1.

	Application A		Application B	
C/I	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
no errors	better than G.722-48k	G.722-56k	G.722-56k	G.722-64k
19 dB	better than G.722-48k		G.722-56k	
16 dB	G.722-48k		G.722-48k	
13 dB	G.722-48k		G.722-48k	
< 13dB	See Note 1		See Note 1	

Table 1a: Clean speech requirements under static test conditions for Applications A and B.

	Application A		Application B	
C/I	**Performance requirement** (see Note 2)	**Performance objective**	**Performance requirement** (see Note 2)	**Performance objective**
no errors	G.722-48k with [10%] PoW	G.722-56k	G.722-56k with [10%] PoW	G.722-64k
19 dB	G.722-48k with [10%] PoW		G.722-48k with [10%] PoW	
16 dB	G.722-48k with [10%] PoW		G.722-48k with [10%] PoW	
13 dB	G.722-48k with [10%] PoW		G.722-48k with [10%] PoW	
< 13dB	See Note 1		See Note 1	

Table 1b: Background noise requirements under static test conditions for Applications A and B.

Notes to Tables 1a and 1b:

Note 1: The AMR WB performance requirement for C/I values below 13dB is the following for Applications A and B: the degradation in subjective performance with each 3dB reduction in C/I shall not be greater than the degradation in subjective performance demonstrated by EFR over the same C/I interval. The specific intervals of interest are 13dB to 10dB, 10dB to 7dB, and 7dB to 4dB. [The test methodology for this requirement is FFS by SQ.]

Note 2: “with 10% PoW” shall be interpreted as no more than 10 additional percentage points of annoying degradation, in terms of annoying or very annoying (i.e. 1+2 votes), with respect to the reference codec. For example, consider a data set where we see that the reference codec has 12 of 344 votes in the annoying or very annoying categories. Thus, the observed proportion of annoying degradation is 0.03, leading to a criterion of a proportion of no more than 0.13 for the codec under test. Suitable statistical methods will be employed. Note that the average DMOS score is not part of this requirement.

2.2. Dynamic conditions

Dynamic conditions refer to channel cases where shadowing is present. Specifically derived channel profiles with varying C/I or C/N will be used.

The requirements for the TCH-FS 22.8 kbit/s traffic channels (applications A and B) under dynamic test conditions are specified in Table 2.

TCH-FS Full-Rate Channel	
Requirement for typical C/I conditions	Better than the EFR under the same conditions
Requirement for difficult C/I conditions ([typical conditions -6dB])	Same or better than the EFR under the same conditions

Table 2: Requirements under dynamic test conditions for Applications A and B

2.3. Additional speech codec performance requirements and objectives

The reference speech codecs for Applications A and B under tandeming, talker dependency, level dependency and language dependency conditions are specified in Table 3.

Tandeming performance and level dependency will be evaluated in the selection phase. It is anticipated that the other additional requirements will be evaluated in the characterisation phase.

	Application A		**Application B**	
Condition	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
Tandeming for clean speech signals (2 asynchronous encodings)	G.722-48k with 2 asynchronous encodings	G.722-56k with 2 asynchronous encodings	G.722-56k with 2 asynchronous encodings	G.722-64k with 2 asynchronous encodings
Low level input speech (-36dBov nominal input level)	better than G.722-48k with -36dBov nominal input level	G.722-56k with -36dBov nominal input level	G.722-56k with -36dBov nominal input level	G.722-64k with -36dBov nominal input level
High level input speech (-16dBov nominal input level)	better than G.722-48k with -26dBov nominal input level	G.722-56k with -26dBov nominal input level	G.722-56k with -26dBov nominal input level	G.722-64k with -26dBov nominal input level
Talker dependency	G.722-48k		G.722-56k	
Language dependency	G.722-48k		G.722-56k	

Table 3a: Additional performance requirements for clean speech signals for Applications A and B

	Application A		Application B	
Condition	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
Tandeming for speech signals with background noise (2 asynchronous encodings)	G.722-48k with 2 asynchronous encodings	G.722-56k with 2 asynchronous encodings	G.722-56k with 2 asynchronous encodings	G.722-64k with 2 asynchronous encodings

Table 3b: Additional performance requirements for speech signals with background noise for Applications A and B

	Application A		Application B	
Condition	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
Tandem with G.711	GSM EFR	Better than GSM EFR	GSM EFR	Better than GSM EFR
Tandem with GSM EFR	GSM EFR with 2 asynchronous encodings	Better than GSM EFR with 2 asynchronous encodings	GSM EFR with 2 asynchronous encodings	Better than GSM EFR with 2 asynchronous encodings

Table 3c: Additional performance requirements for tandeming with a narrowband system for Applications A and B

Notes to Table 3c:

Note 1: These conditions will be tested for both tandem configurations, i.e. the narrowband codec preceding the wideband codec and *vice versa*.

Note 2: An appropriate testing methodology for these conditions is to be determined. One option is to include them in a narrowband-only experiment in the selection tests, which may already be needed for testing reference coders for low C/I ratios. At the very least, codec proponents may be asked to include these conditions in demo material to be submitted as part of the stage 2 deliverables.

3. Requirements and Objectives for Applications C, D and E

3.1. Performance with channel errors

The performance requirements and objectives for Applications C and D with channel errors are specified in Table 4; the performance requirements and objectives for Application E with channel errors are specified in Table 5. The performance requirements for music are provided in Table 6.

	Application C		Application D	
C/I	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
no errors	G.722-64k		G.722-64k	
19 dB	G.722-64k		G.722-64k	
16 dB	G.722-56k	G.722-64k	G.722-56k	G.722-64k
13 dB	better than G.722-48k	G.722-56k	better than G.722-48k	G.722-56k
< 13dB	See note		See note	

Table 4a: Clean speech requirements under static test conditions for Applications C and D. Application D assumes n=2 (i.e. 45.6 kbps)

	Application C		Application D	
C/I	**Performance requirement**	**Performance objective**	**Performance requirement**	**Performance objective**
no errors	G.722-56k	G.722-64k	G.722-56k	G.722-64k
19 dB	G.722-56k		G.722-56k	
16 dB	G.722-48k		G.722-48k	
13 dB	G.722-48k		G.722-48k	
< 13dB	See note		See note	

Table 4b: Background noise requirements under static test conditions for Applications C and D. Application D assumes n=2 (i.e. 45.6 kbps)

Notes to Tables 4a and 4b:

Note: The AMR WB performance requirement for C/I values below 13dB is the following for Applications C and D: the degradation in subjective performance with each 3dB reduction in C/I shall not be greater than the degradation in subjective performance demonstrated by EFR over the same C/I interval. The specific intervals of interest are 13dB to 10dB, 10dB to 7dB, and 7dB to 4dB. [The test methodology for this requirement is FFS by SQ.]

	Application E (see note 1)	
EC / [FER, RBER] (see note 2)	**Performance requirement**	**Performance objective**
No errors (see note 3)	G.722-64k	
[0.5%, 0.0%],	G.722-56k	
[1.0%, 0.1%], note 4, UL	G.722-48k	
[1.0%, 0.1%], note 4 , DL	G.722-48k	
[1.0%, 0.1%] note 5, UL		G.722-48k

Table 5a: Clean speech under channel errors for Application E.

	Application E (see note 1)	
EC / [FER, RBER] (see Note 2)	**Performance requirement**	**Performance objective**
No errors (see note 3)	G.722-64k	
[0.5%, 0.0%],	G.722-56k	
[1.0%, 0.1%], note 4, UL	G.722-48k	
[1.0%, 0.1%], note 4 , DL	G.722-48k	
[1.0%, 0.1%] note 5, UL		G.722-48k

Table 5b: Background noise requirements under channel errors for Application E.

Notes to table 5a and 5b:

Note 1: Application E includes all bit rates. The requirements are however only tested for the highest modes

Note 2: The error performance for Application E is specified and evaluated using error protection schemes from the UTRAN toolbox. Each error condition (EC) is defined using two error profiles, one FER profile (single indicator per frame) and one residual BER profile (bit-level residual error channel). Both profiles are derived from WCDMA channel simulations with the following parameters:

- Maximum source bitrate is [32 kbit/s], errored frames of size [20 ms] will be used
- Spreading factor is [64]

- CRC size class a is [16 bits]
- Urban outdoor channel profile
- 3 km/h UE speed
- TFCI bits included
- Normal frames (not compressed)
- No DL transmitter diversity
- 2 pilot bits
- One gain factor

Note 3: The requirement for the no error case applies to modes with higher bit rates, i.e. not tested in applications A and B

Note 4: The least significant bits shall be subjected to the residual error profile. The number of bits in this class shall be 25% of the total bits per frame.

Note 5: The least significant bits shall be subjected to the residual error profile. The number of bits in this class shall be 40% of the total bits per frame.

Condition	Requirement	Objective
Music	No annoying effects	G.722-56k

Table 6: Requirements and objectives with music for Applications C, D and E

3.2. Additional speech codec performance requirements and objectives

The reference speech codecs for Applications C, D and E under tandeming, talker dependency, level dependency and language dependency conditions are specified in Table 7.

	Applications C, D and E	
Condition	**Performance requirement**	**Performance objective**
Tandeming for clean speech signals (2 asynchronous encodings)	G.722-64k with 2 asynchronous encodings	
Low level input speech (-36dBov nominal input level)	G.722-64k with -36dBov nominal input level	
High level input speech (-16dBov nominal input level)	G.722-64k with -26dBov nominal input level	
Talker dependency	G.722-64k	
Language dependency	G.722-64k	

Table 7a: Additional performance requirements for clean speech in Applications C, D and E

	Applications C, D and E	
Condition	**Performance requirement**	**Performance objective**
Tandeming for speech signals with background noise (2 asynchronous encodings)	G.722-56k with 2 asynchronous encodings	

Table 7b: Additional performance requirements for speech with background noise in Applications C, D and E

4. Requirements and Objectives for All Applications

The performance requirements and objectives under bit-rate switching and DTX are specified in Table 8; the performance requirements and objectives for DTMF, information tones and idle noise are specified in Table 9.

Condition	Requirement	Objective
Switching between different AMR-WB bit-rates	No annoying artefacts	
Clean speech with DTX enabled	Performance with DTX disabled	
Speech and background noise with DTX enabled	Performance with DTX disabled	

Table 8: Additional performance requirements for speech signals (all applications)

Condition	Requirement	Objective
DTMF		Transparent transmission of DTMF.
Information tones	Recognisable as given information tone.	
Idle noise	-66dBm0 (unweighted)	

Table 9: Requirements and objectives for speech codec performance with non-speech inputs

5. Open Issues

This section lists open issues currently under discussion.

- Performance in tandem with other standards:
    - G.722 (selection and/or characterisation phase)
    - Other WB standards
- Performance under mode switching between NB and WB AMR
- Performance definition and testing for application E (and during which phases these are to be addressed)

Document History

Version	Date	Comment
0.1	October 1999	Initial version
0.2	October 1999	ETSI-SMG11#12/3GPP-SA4#7
0.3	December 1999	ETSI-SMG11#13/3GPP-SA4#8
1.0	December 1999	3GPP-SA
1.1	January 2000	ETSI-SMG11#14/3GPP-SA4#9

Joint 3GPP TSG-S4#9 – SMG11#14 meeting ***Tdoc S4/SMG11 87/00***
January 24-24, 2000, Puerto Vallarta, Mexico

Title:	**Permanent project document WB-4: Design Constraints, v.1.0**
Source:	**S4/SMG11**
Document for:	**Information**
Agenda Item:	**16**

1. Introduction

This document lists design constraints for the AMR-WB speech codec development. The design constraints are a set of mandatory requirements that the codec proposal must fulfil to be suitable for 3G and GSM and to be considered in codec selection. Whenever it is not clearly specified, the constraints apply to the selection phase and to any preselection (qualification) phase of the AMR-WB development.

2. AMR-WB codec development constraints (summary table)

<table>
<tr><th colspan="2">Development constraints</th><th>Open issues, notes</th></tr>
<tr><td>Complexity requirements[1]</td><td></td><td>WB = Wideband
NB = Narrowband</td></tr>
<tr><td>Channel coding including possible control loop management algorithms</td><td><u>GSM FR:</u>
A. wMOPS ≤ 5.7 wMOPS
B. RAM ≤ 3.0 kwords
C. ROM ≤ 4.5 kwords
D. Program ROM ≤ 1.5 * Program ROM of AMR-NB FR ch. Codec) (1.5 * 1 342 ETSI basic operators)
<u>GSM EDGE:</u>
[t.b.a.]
<u>GSM multi-slot:</u>
[f.f.s.]
<u>3G:</u>
Existing generic 3G channel coding toolbox shall be used for channel coding.</td><td>Are separate (higher) complexity requirements needed for higher rate GSM channels?</td></tr>
<tr><td>Speech coding (excluding VAD/DTX)</td><td>E. wMOPS ≤ 40 wMOPS (≈ 2.4 x wMOPS of AMR-NB sp. Codec: 16.75)
F. RAM ≤ 15 kwords (≈ 2.8 × RAM of AMR-NB speech codec: 5.28 kwords)
G. ROM ≤ 18 kwords (≈ 1.2 x ROM of AMR-NB speech codec: 14.57 kwords)
H. Program ROM ≤ 1.2*Program ROM of AMR-NB speech codec (= 1.2 * 4 851 ETSI basic operators)</td><td>The complexity limit applies to the codec considered as a whole, including all modes.</td></tr>
<tr><td>Additional complexity for VAD/DTX operation (over speech coding complexity limits)</td><td>1.5 times the corresponding complexity of AMR-NB VAD/DTX with the more complex VAD Option (VAD2):
I. wMOPS ≤ 1.6 wMOPS
J. RAM ≤ 149 words
K. ROM ≤ 1004 words
L. Program ROM ≤ 1.5 * Program ROM of AMR-NB VAD/DTX with the more complex VAD Option (VAD 2) ([t.b.a.] ETSI basic operators)</td><td></td></tr>
<tr><td>A-ter and Iu submultiplexing</td><td>At least one codec mode at AMR-WB shall be consistent with 16 kbit/s submultiplexing on the A-ter interface. This implies the constraint of providing at least one codec mode in AMR-WB operating at a source codec bit-rate below 14.4 kbit/s .</td><td>Bit-rate limitations due to EDGE A-ter?</td></tr>
</table>

Other constraints for bit-rates	The source codec shall be capable of operating at or below the primary rates of interest to ITU WB Question (currently bit-rates of 16 and 24 kbit/s).	Investigate the possible constraints due to multi-slot operation.
Codec mode	**GSM FR:** • Same signalling scheme as in AMR-NB shall be used. This is valid for codec mode and channel measurement signalling. **GSM EDGE:** • Same signalling scheme as in AMR-NB shall be used. This is valid for codec mode and channel measurement signalling. **GSM multi-slot:** • Same signalling scheme as in AMR-NB shall be used. This is valid for codec mode and channel measurement signalling. This constraint is to be confirmed at the next meeting. **3G:** • Same signalling scheme as in AMR-NB shall be used. Channel coding (including possible control-loop management algorithms) is part of the codec proposal. The channel coding scheme shall be included for GSM FR, GSM EDGE, GSM multi-slot and 3G.	Note that there are no constraints for the number of codec modes. Nevertheless the number of codec modes allowed for each channel in selection/qualification tests may be restricted (see WB-8 "Test Plans for Each Phase"). Availability of error patterns must be confirmed.

<table>
<tr><td>Channel mode</td><td><u>GSM (FR, EDGE and multi-slot):</u>

The AMR-WB codec will operate in GSM full-rate speech traffic channel, EGDE, and multi-slot channels.

Channel mode handovers will be executed in the same way as existing intra-cell handovers. Handovers between AMR-WB FR and AMR-NB HR will mean switching between wideband speech services and the existing AMR HR narrowband speech services. The algorithm used to determine when and whether to perform an AMR handover will be specific to the BSS manufacturer.

• Channel mode signalling: transmitted out of band on the radio interface
• The up- and downlinks (of the same air-interface) shall use the same channel mode.
• Channel mode control is located in the network.</td><td><u>3G:</u>

The AMR-WB will operate with spreading factors: 128, 64 and 32 (see 3G channel codec toolbox)</td><td></td></tr>
<tr><td>Channel coding</td><td><u>GSM (FR, EDGE and multi-slot):</u>

The existing sets of convolutional polynomials defined in GSM 05.03 shall be used.</td><td><u>3G:</u>

Existing generic 3G channel coding toolbox shall be used.

Error protection containing up to 3 bit-sensivity classes may be used.</td><td></td></tr>
<tr><td>Tandem Free Operation (TFO)</td><td colspan="2">The AMR-WB codec shall support Tandem Free Operation</td><td></td></tr>
<tr><td>Voice Activity Detection (VAD) and comfort noise</td><td colspan="2">The codec proponents shall provide the VAD and comfort noise encoding solution for the selection phase. The AMR-WB VAD/DTX will be that associated with the selected codec.</td><td></td></tr>
<tr><td>Discontinuous Transmission (DTX)</td><td><u>GSM FR:</u>

The same DTX scheme (transport format, update frequency) as in AMR-NB shall be used

<u>GSM EDGE:</u>

Derived from the AMR-NB DTX

<u>GSM multi-slot:</u>

Derived from the AMR-NB DTX</td><td><u>3G:</u>

The same SCR-scheme (transport format, update frequency) as in AMR-NB shall be used.</td><td></td></tr>
</table>

Active noise suppression in the selection/ qualification phase	In order to compare all solutions in the same conditions, and select the candidate with the best intrinsic quality, the noise suppressers would not be included during the selection phases, or that any noise suppresser integrated to a source codec shall be turned off for these tests. The selection and possible standardisation of a noise suppresser may then be addressed in a separate phase	
Transmission delay[2]	This constraint is set for the algorithmic transmission delay in GSM FR channel. The target is to keep the algorithmic round trip delay for wideband modes equal to the algorithmic round trip delay of the GSM AMR-NB FR. Nevertheless, some increase of algorithmic transmission delay is expected due to the higher source coding bit-rates in AMR-WB. See note 2 "Evaluation of algorithmic round-trip delay" for definition of the evaluation methodology and the relating constraint.	
Error concealment	Error concealment techniques of AMR-WB codec candidates shall only rely on soft-output information from the equaliser (in BTS only information that can be sent over Ater). This does not preclude any future exploitation of other radio channel parameters in the final AMR-WB system.	
Frame size	The frame size is constrained to be one of the possible values: 5ms, 10ms or 20 ms.	
Input sampling rate and audio bandwidth	The codec will operate on 16 kHz input sampling rate. The input signal bandwidth shall be 50 Hz to 7 kHz. It is required to make sure that no artifacts are caused by signals lying outide the range 50 Hz to 7 kHz.	Note that other bandwidths may be used for testing purposes.

[1] The complexity requirements are valid for all phases of the AMR-WB development and are separate for channel coding, speech coding and DTX algorithms.

Notes:

1. Program ROM is computed as the number of basic instructions
2. The control loop management algorithms are intended to include all the additional algorithms beyond speech and channel codec that are needed for codec mode adaptation: channel metric estimation, adaptation algorithm, coding and decoding of the in-band signalling.

Complexity calculation rules:

The same complexity evaluation methodology as used in the past for GSM AMR narrowband standardisation (based on ETSI fixed-point basic operations) will be used for complexity evaluation of the AMR-WB codec. Detailed procedure for each phase is the following:

- **Qualification (if needed)**: Complexity evaluation may be based on floating point code. The results should nevertheless be presented as ETSI FOM, wMOPS, and memory figures even though they are allowed to be estimated from a floating-point code. Requirements shall be checked according to the assessment methodology given in AMR narrowband document AMR-9 (Complexity and delay assessment) [2].
- **Selection**: ETSI methodology based on fixed point code (Basic op. Counters, i.e. Worst observed case)
- **Verification/characterisation**: ETSI methodology based on fixed point code (Theoretical worst case)

Arithmetic used in codec proposals:

- **Qualification (if needed)**: Fixed point or floating point code
- **Selection**: Fixed point code (using ETSI set of basic operations)
- **Verification/characterisation**: Fixed-point code (using ETSI set of basic operations).

2 Evaluation of algorithmic round-trip delay

The MS-to-MS algorithmic round-trip delay evaluation of a codec mode is based, as for the GSM HR, EFR and AMR standardisation, on four codec dependent algorithmic delay contributors :

- analysis frame length delay (T_{sample}): duration of the segment of PCM speech operated on by the speech transcoder.
- interleaving and de-interleaving delay (T_{rftx}): time required for transmission of a speech frame over the air interface due to interleaving and de-interleaving.
- uplink Abis delay (T_{Abisu}): time needed to transmit the minimum amount of bits over the Abis interface that are required at the speech decoder to synthesise the first output sample.
- downlink Abis delay (T_{Abisd}): time required to transmit all the speech frame data bits over the Abis interface in the downlink direction that are required to encode one speech frame.

The formula used for round-trip delay evaluation is the following:

$$D_{round\text{-}trip} = 2(T_{sample} + T_{rftx}) + T_{Abisu} + T_{Abisd}$$

The proponents must compute and provide figures for each component of the round trip delay for following configurations :

- highest delay of full-rate modes with 16kbps sub-multiplexing scheme ;
- highest delay of full-rate modes without 16kbps sub-multiplexing scheme

The delay constraints for the GSM TCH/FR channel is defined as follow:

Let's define the algorithmic round trip delay with two components

$$D_{round\text{-}trip} = 2(T_{sample} + T_{rftx}) + T_{Abisu} + T_{Abisd}$$

$$D_{round\text{-}trip} = D_{rt1} + D_{rt2}$$

With

$D_{rt1} = 2(T_{sample} + T_{rftx})$ **,** the algorithmic round trip delay without the Abis-Ater interface component, and

$D_{rt2} = T_{Abisu} + T_{Abisd}$ **,** the algorithmic round trip delay component over the Abis-Ater interface.

The reference are the maximum D_{rt1} and D_{rt2} delays for the AMR narrow-band. I.e. taken from the GSM 06.75 v7.1.0 we have :

D_{rt1} (ref)= $2(T_{sample} + T_{rftx})$ = 125 ms for FR-12.2 kbps AMR mode

D_{rt2} (ref)= $T_{Abisu} + T_{Abisd}$ = 24.25 ms for FR-12.2 kbps AMR mode

All AMR WB candidates shall comply with the following:

D_{rt1} (WB) <= D_{rt1} (ref) ; for scenarii A (GSM full-rate traffic channel (22.8 kbit/s gross bit-rate) with an additional constraint of 16 kbit/s A-ter sub-multiplexing) and B (GSM full-rate traffic channel (22.8 kbit/s gross bit-rate), and

D_{rt2} (WB) <= D_{rt2} (ref) + 5 ms ; for scenario A.

The Abis delays (uplink and downlink) should be computed with a similar methodology as the one used in the GSM Recommendations 03.05 for the GSM FR, and 06.55 for the GSM EFR.

For the qualification and the selection phases the proponents must justify how the delay figures were computed. To that purpose, they can base the Abis delays on either type of TRAU frames (existing FR, HR and AMR TRAU frames, or new TRAU frames format proposed for AMR-WB).

REFERENCES

[1] "Adaptive Multi-Rate Wideband (AMR-WB) Feasibility Study Report", version 1.0.0, Source: SMG11, ETSI TC SMG Tdoc SMG P-99-429, Meeting #29, 23-25 June, 1999, Miami (FL), USA

[2] "Reproduction of AMR narrowband document AMR-9 (Complexity and delay assessment)", v1.3, Tdoc S4/SMG11 71/00.

DOCUMENT HISTORY

VERSION	SOURCE
v0.1	Initial version by editor (October 1999) Editor: Kari Järvinen, Nokia; Mailing address: Nokia Research Center, P.O. Box 100 (Visiokatu 1), FIN-33721 Tampere, Finland; Email: kari.ju.jarvinen@nokia.com; Tel:+358 3272 5854; Mobile: +358 50 555 0 999; Fax: +358 3272 5888
v0.2	AMR-WB subgroup during SMG11#12 (October 1999)
v0.3	SMG11#12 (October 1999)
v0.4	editor (December 1999)
v0.5	AMR-WB subgroup at SMG11#13, approved at SMG11#13 (December 1999)
v0.6	AMR-WB subgroup at SMG11#14 (January 2000)
v1.0	SMG11#14 (January 2000)