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Contents

Foreword

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where:

x the first digit:

1 presented to TSG for information;

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y the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.

z the third digit is incremented when editorial only changes have been incorporated in the specification;

Introduction

This document contains the AMR Wideband Feasibility Report produced by 3GPP TSG-S4 and ETSI SMG11 before the approval of the related Work Item by TSG-SA.

Executive Summary and Recommendations

Introduction

As asked by SMG in October 1997, SMG11 has conducted a study into the feasibility of AMR wideband (AMR-WB) speech codec. This study addresses not only the technical feasibility but also the benefits of AMR wideband, the development plan and Timescales.

Benefits

While current GSM codecs achieve good performance for narrowband speech (audio bandwidth limited to 3.4 kHz), the introduction of a wideband speech service (audio bandwidth extended to 7 kHz) would provide improved voice quality especially in terms of increased naturalness in voice. Wideband coding would bring speech quality exceeding that of (narrowband) wireline quality to GSM. There is a growing market interest in a wideband speech service among operators and in having an adaptive multi-rate wideband codec included into GSM AMR.

Performance

The AMR-WB is operable on several channels, including existing and evolved GSM channels. The operation on a GSM full-rate single slot channel is seen as one application, in which case the AMR-WB coder will provide:

- Improved speech quality (exceeding G.711 PCM wireline quality) for medium and low error rates (down to 13 dB C/I) through the introduction of the wider audio bandwidth (7 kHz). The quality for low error conditions will be comparable to 56 kbit/s wideband speech in ISDN.
- Channel robustness, i.e. performance at lower C/I, similar to that of the existing narrowband GSM FR and GSM EFR speech coders

In addition, AMR-WB will be applicable also to other higher rate channels (e.g. EDGE), in which case further improvements in quality are expected since higher source coding rates are possible to use.

Risk areas

The main performance limitations and technical risk areas have been identified as follows:

- Codec performance – performance in poor channel conditions: AMR-WB is primarily intended for low and medium error-rate channel conditions (due to the need for rather high source coding bit-rates). AMR-WB should provide a level of satisfactory performance also in poor transmission conditions. During codec development phase, care should be taken to provide also sufficient error resilience for the AMR-WB.
- Architectural and network issues – The signaling for the establishment of a wideband call has to be defined. The format of the wideband speech on the A-interface needs to be further studied. This can have an influence on the design constraints. It is therefore important that the work item is properly managed by the different involved STCs.

Recommendations

On basis of the performance benefits and the risks highlighted above, the following recommendations are submitted to SMG for approval:

- Initiate a program to develop, test, select and specify AMR wideband (AMR-WB) together with related features such as VAD/DTX.
- An acceptable and workable timeplan should be agreed by SMG11 (and SMG1/SMG2/SMG3/SMG12) as soon as possible after SMG#29 with the priority of targeting approval of AMR-WB codec by end of year 2000 for GSM Release 2000.
- Approve Work Item description in Annex F

Introduction

The AMR wideband (AMR-WB) service option was considered within SMG11 already during the AMR narrowband feasibility study and development phases during 1997-1998. Many aspects of wideband were not resolved at that time due to lack of time and conflicting opinions. SMG meeting #23 (October 1997) decided that the AMR narrowband program should not be delayed for the sake of introducing a wideband mode and that feasibility assessment with regard to wideband should be completed as soon as possible. Wideband could then be added to AMR as a later option according to the outcome of the feasibility assessment. Due to intensive AMR narrowband development and standardization, the main part of the wideband feasibility study has been carried out after SMG meeting #27 (October 1998).

Scope

This report presents the outcome of the Feasibility Study Phase of wideband Adaptive Multi-rate codec. Sections 1 to 3 provide an overview and background for introducing AMR wideband. Section 4 provides a review of relevant standardization activities. Feasibility aspects such as codec performance, speech coding bit-rates, implementation complexity and transmission delay are discussed in sections 5 and 6. Section 7 outlines the target timeplan for AMR-WB codec development, testing, selection and standardization. Sections 8,9 and 10 conclude with a summary of the risks and recommendations.

1. Goals of AMR wideband codec

The goals of the AMR-WB are to develop wideband speech coding targeted for existing and evolved GSM systems as well as other related mobile systems, such as the 3GPP system. The AMR-WB coding will provide:

- In medium and low error conditions; improved speech quality (exceeding G.711 PCM wireline quality) through the introduction of the wider audio bandwidth (7kHz)
- A wideband speech service with a channel robustness similar to that of the existing narrowband GSM FR and GSM EFR speech coders when used in GSM
- A wideband speech service which will be applicable to existing and evolved GSM systems as well other mobile systems, thus enabling efficient wideband speech service interoperation
- Added flexibility to AMR, which can be used to tailor the speech services to specific operator needs

2. Terminology

The terminology and acronyms used in this report are given in Annex A.

The current feasibility study report extensively uses AMR, AMR-NB and AMR-WB abbreviations. AMR-NB corresponds to the AMR narrowband speech service, which has been recently standardized for GSM. AMR-WB is the wideband speech service, based on an adaptive multirate coding scheme, studied in this report. AMR is the generic name used to identify the adaptive multirate concept as well as the set of speech services for GSM, and possibly other systems, based on this concept.

3. Overview

3.1 Benefits

The current GSM speech codecs (FR, HR, EFR and AMR) operate for narrowband speech (audio bandwidth limited to 3.4 kHz). While these codecs achieve good performance for narrowband speech, the introduction of a wideband speech service (audio bandwidth extended to 7 kHz) would provide improved voice quality and increased naturalness of voice, especially in handsfree situations.

A wideband service would give a substantial quality improvement over narrowband service and it is foreseen to be a market for mobile-to-mobile wideband users (or mobile-to-fixed), although the service will initially be restricted by the number of users having the necessary wideband terminals.

There is a growing market interest in a wideband speech service among operators and in having a wideband codec included into GSM AMR.

3.2 Basic operation

AMR-WB consists of one or several codec modes, with the possibility to switch between the different codecs as a function of channel error conditions following the principles of AMR. (See [1] and [2] for functional description of AMR.) Each wideband codec mode provides a different level of error protection through a different distribution of the available gross bit-rate between source and channel coding. The most robust wideband codec mode is chosen in poor channel conditions, while the codec mode providing best clean-channel quality is chosen in good channel conditions.

The AMR-WB coder will operate in similar way as AMR-NB, using channel quality measurements and an in-band signaling channel to adapt the modes. The coder will be equipped with a VAD/DTX system in analogy to the VAD/DTX for the narrowband coders. Similarities to existing AMR should be preferred for compatibility and implementation reasons unless performance improvement can be obtained.

Development focus will be on the GSM full-rate traffic channel (22.8 kbit/s). At least one of the codec modes shall be compatible with 16 kbit/s sub-multiplexing. Also higher source coding bit-rate modes are considered to be used with higher sub-multiplexing bit-rate over A-ter and in anticipation of future mobile channels that could utilize higher bit-rates, especially GSM EDGE speech channels and also 3G channels.

3.3 Application scenarios

The AMR-WB coder will be operable on several mobile channels including existing (single slot full-rate) and evolved (EDGE, multi-slot) GSM systems as well as other related mobile systems, such as the 3GPP system. Wideband coding is expected to require relatively high source coding bit-rates (above 10 kbit/s) which may not be feasible in lower rate channels such as the GSM half-rate channel. The focus is therefore on the GSM full-rate channel and other higher rate mobile channels.

For the low and medium rate AMR-WB modes, the focus is primarily on wideband speech and not wideband music. For the higher rate modes, improved performance also for non-speech signals, e.g. music, is expected, but this is for further study.

For GSM, AMR-WB is intended to be used in low to medium error-rate channel conditions where it provides performance benefits over AMR narrowband. In high error conditions, degradation of AMR-WB is comparable to the degradation in EFR. In poor operating conditions, it may be reasonable to switch to use the AMR narrowband, which enable more bits for error protection and better error resilience.

A wideband call can be established only when both ends have wideband capable terminals. This requires that wideband capable audio parts are introduced in terminals, and is likely to imply that the use of AMR-WB will, at least initially, be restricted to MS-to-MS calls.

3.4 Development Timescales

There is a growing market interest in a wideband speech service among operators and in having a wideband codec mode(s) included into GSM AMR. The target for the development is set for approval of AMR-WB codec in SMG#33 on 6-10 November, 2000, and approving AMR-WB specifications in SMG#34. AMR-WB would be introduced in GSM Release 2000.

4. Standardization background

4.1 GSM

The ETSI have recently standardized different new schemes for increasing the bit-rate available to the mobile users (e.g. HSCSD, EDGE, and GPRS). The AMR standardization has shown the interest of the multi-rate approach to get the best trade-off in term of quality and capacity. A logical evolution is to offer such multi-rate codecs on the new transmission schemes. This must be taken into account when standardizing a WB speech service for the GSM.

It seems that in the near future there will be multiple mobile channels that could utilize wideband coding at higher bit-rates. These may include EDGE and 3G channels, and possibly even data channels such as HSCSD and GPRS. Higher bit-rates could be facilitated also by multi-slot solution. It seems evident that a multi-rate wideband codec solution should be developed in anticipation of new channels and compatibility requirements between these systems. Development of higher bit-rate codecs requires close collaboration with SMG STCs and 3GPP.

4.1.1 EDGE

EDGE Phase 1 standard will be in Release 99 of GSM specifications. It contains enhancements for GSM data services HSCSD and GPRS (ECSD and EGPRS, respectively). EDGE speech services could be defined in EDGE Phase 2 standardization, targeting for Release 2000. Defining speech services over circuit switched 8-PSK modulated EDGE channels has the benefits of improved speech quality through higher bit-rate wideband codecs.

On the SMG2 #31 meeting (31st May – 4th June), EDGE phase 2 has been discussed in more detail. SMG2 has received proposals, which have identified several different services and applications as possible candidates to be supported by EDGE phase 2. These services include high quality/capacity speech service using the AMR wideband codec or the usage of half- or quarter-rate channels using the existing AMR codecs.

It is planned to start the work for EDGE phase 2 in SMG2 immediately and to target the possible changes for release 2000. SMG2 has sent a liaison statement to SMG asking to propose a way to accomplish the service and radio requirements for EDGE phase 2 and the way forward.

According to the defined EDGE frame and burst structure, the following new speech bearers could be defined in EDGE phase 2.

Table I. EDGE bearers and higher bit-rates

Speech Channel Mode	Gross bit-rate per channel	Note
EDGE Full-rate channel	68.4 kbit/s	
EDGE Half-rate channel	34.2 kbit/s	
EDGE Quarter rate channel	17.1 kbit/s	
EDGE Eighth rate channel	8.55 kbit/s	This mode may not be possible due to the small interleaving depth.

Table I shows that EDGE bearers would provide new possibilities to include higher bit-rates for wideband speech coding. The maximum gross bit-rate over the air-interface is 68.4 kbit/s. In the network, the maximum transmission bit-rate per single time slot is restricted to 64 kbit/s.

GSM evolution (e.g. EDGE) provides possibilities to use higher bit-rates and these should be considered in the current AMR wideband speech codec development.

4.2 3GPP

A work item for the standardization of a wideband speech codec has been created in 3GPP TSG-SA Working Group 4 (Speech codec) for “Codec(s) for Wideband Telephony Services” (WI S4). The technical scope of this work item is to consider existing standard codec(s), results of on-going standardization work in other standardization bodies and new wideband codec(s) with the objective to select the best possible solution for the wideband telephony services within the 3G mobile telephony framework. Especially, the results of the wideband speech codec standardization work in ITU-T, ISO-MPEG, ETSI SMG11 and ARIB should be considered. E.g., 3G has noted that ETSI SMG11 has scheduled to complete the feasibility phase on AMR wideband in June 1999.

Schedule of tasks to be performed:

- Approval of WI: TSG SA #2 (March 1999)
- First draft of requirements: June 1999
- Final definition of requirements: October 1999
- Selection of codecs: April 2000
- Baseline Specifications: (TBD)
- Codec characterization in 3G radio channels: (TBD)
- Final specification: (TBD)

4.3 ITU-T

Since 1996, ITU-T Question 20 / Study Group 16 has been performing standardization activities for a wideband coding scheme. The original question at ITU-T in 1995 did ask for a wideband-coding algorithm working at three bit-rates: 16 kbit/s, 24 kbit/s and 32 kbit/s. the coding was intended for speech and music for all three bit-rates. However, subjective tests indicated that it was not possible to achieve the quality for music at the lowest bit-rate mode (16 kbit/s). Therefore, in 1998, a new activity was started in order to find a coding scheme working at 24 kbit/s and 32 kbit/s only. Formally, the question for a wideband coding scheme working at 16 kbit/s is now completely separated from the higher bit-rate solutions.

In May 1999, results for two codec candidates for 24 kbit/s and 32 kbit/s were considered in Study Group 16. Two codec candidates had been tested and a selection between these was made. (All requirements were not met by either of the tested codecs.) Also, a new project activity "wideband (7 kHz) speech coding algorithm around 16 kbit/s" was launched. This new activity is focused on speech signals.

The question of ITU-T's experience in the wideband issue has been addressed during the AMR-WB feasibility study [e.g., 8]. Considering the broader range of applications foreseen for the new ITU-T wideband codec, this codec might not necessarily be optimized or a total solution for mobile applications.

4.3.1 ITU-T new wideband activity around 16 kbit/s

In May 1999, the following guidelines have been considered relevant in ITU-T for the new wideband activity around 16 kbit/s (12, 16, 20, and 24 kbit/s):

- Input and output audio signals should have a bandwidth of 7 kHz at a sampling rate of 16 kHz.
- Primary signals of interest are clean speech and speech in background noise. Music performance requirements set at higher bit-rates (24 kbit/s).
- High speech quality with the objective of equivalence to G.722 at 56/64 kbit/s.
- 16 kbit/s is the main bit-rate. It is required that the ability of the candidate to scale in bit-rate to lower bit-rates (less then 16 kbit/s) and up to 24 kbit/s with no fundamental changes in either the technology or the algorithm used.
- Robustness to frame erasures and random bit errors.
- Low algorithmic delay (frame size of 20ms or integer sub-multiples)

The applications for the new activity were considered as follows: Voice over IP (VoIP) and Internet Applications, PSTN applications, Mobile Communications, ISDN wideband telephony, and ISDN videotelephony and video-conferencing [12].

With regard to the timeschedule, it is generally agreed that if frozen Terms of Reference and corresponding Qualification test plan are ready at the next Rapporteur's meeting (September 1999), the possibility to have To time (starting point in time of the standardization process) at the next SG16 meeting (February 2000) will be considered.

As an example, the performance requirements set for clean speech conditions are summarized in Table II (taken from [12]; nominal level of –26 dBov).

Table II. ITU-T Performance requirements and objectives for a wideband (7 kHz) speech coding algorithm around 16 kbit/s.

Parameter	Requirement	Objective
Bit-rate(s)	12, 16, 20, 24 kbit/s	more finely scalable
1) at 12 kbit/s	As good as possible	G.722 at 48 kbit/s
2) at 16 kbit/s	Better than G.722 at 48 kbit/s	G.722 at 56 kbit/s
3) at 20 kbit/s	Equivalent to the 16 kbit/s	Better than the 16 kbit/s
4) at 24 kbit/s	G.722 at 56 kbit/s.	G.722 at 64 kbit/s

In addition, at ITU-T / SG16 additional requirements have been specified for transmission errors (specified in terms of bit error rates; BER=10^{-3}), for the conditions of detected frame erasures, and for some other conditions[1], such as background noise and level dependency.

Both ETSI SMG11 and ITU-T/SG16 are currently interested in the standardization of a wideband-coding scheme. The range of bit-rates in both cases includes rates in the vicinity of about 16 kbit/s, which is reasonable for mobile telephone applications. However, ITU-T SG16 does not traditionally cover transmission-related issues such as channel coding. It is felt that besides the random bit error and frame erasure conditions more specific channel models (e.g. channel models for new systems as EDGE, 3GPP) should be used in testing codec performance in order to make them more interesting for mobile applications.

5. Feasibility aspects for GSM full-rate traffic channel

5.1 Basis for the feasibility study

The AMR wideband service option was considered already during the AMR narrowband feasibility study and development phases. Many aspects of wideband were not resolved at that time due to lack of time and conflicting opinions. However, a set of provisional working assumptions was agreed in November 1997 in SMG11 to form the basis for the AMR-WB feasibility study. Table III contains these basis assumptions (reproduced below from [3]).

Table III. Performance objectives for the wideband speech coding algorithm

Condition	Target
Clean Speech, EP0	Not worse than ITU-T Rec. G.722 at 56 kbit/s
Clean Speech, EP1 (10 dB)	$MOS_{WB}(EP0) – MOS_{WB}(EP1) \leq MOS_{EFR}(EP0) - MOS_{EFR}(EP1) + [0.2]$
Clean Speech, EP2 (7 dB)	$MOS_{WB}(EP1) – MOS_{WB}(EP2) \leq MOS_{EFR}(EP1) - MOS_{EFR}(EP2) + [0.3]$
Clean Speech, EP3 (4 dB)	$MOS_{WB}(EP2) – MOS_{WB}(EP3) \leq MOS_{EFR}(EP2) - MOS_{EFR}(EP3) + [0.3]$
Office noise (SNR = 20 dB)	
Babble noise (SNR = 20 dB)	Not worse than ITU-T Rec. G.722 at [56] kbit/s

[1] *It is not possible in the presence of a channel coding scheme to directly compare conditions specified in terms of C/I to conditions specified in terms of BER. We therefore do not repeat the full list of ITU-T's performance requirements here.*

Car noise (SNR = 15 dB)	
Talker Dependency	Not worse than ITU-T Rec. G.722 at 56 kbit/s
Bandwidth	[50] Hz to 7 kHz
Bit-rate	Less than [15] kbit/s for single timeslot operation and 16 kbit/s sub-multiplexing on the A_{ter} interface
Delay	Not more than the GSM EFR round trip delay + 5%
Complexity of combined narrow and AMR wideband	Not more than 8 times the FR complexity
Music	[No annoying artifacts]

It was also acknowledged that depending on the outcome of the feasibility assessments, a new performance requirement specification, possibly with some relaxation, might be proposed for the codec development and selection. A multi-timeslot solution was also to be considered as an option but the emphasis was seen to be on a single timeslot solution.

5.2 Speech quality performance

One of the reasons for assigning some time for the feasibility study was to allow companies to contribute to this phase by performing tests based on prototype solutions for a wideband coding scheme. This was seen as highly desirable in order to get an empirical basis for defining reasonable performance objectives and design constraints. However, since the beginning of the feasibility phase, the number of contributions describing test results related to the wideband issue was very limited ([7, 9, 13]).

In the following sections the results presented within SMG11 contributing to the feasibility study are being summarized. It must be noted that these results are collected from different tests and are not directly comparable to each other.

5.2.1 Benefit of wideband over narrowband

A comparison of the two bandwidths was performed and the results were presented in [7]. In this test, the GSM EFR codec was used as a narrowband coding device. It was compared to the ITU-T Rec. G.722 @ 48 kbit/s and G.722 @ 56 kbit/s wideband codecs.

Table IV. Results for narrowband and wideband speech

Codec	MOS
GSM EFR (narrowband; 3.4kHz)	3.3
G.722 @ 48 kbit/s (wideband; 7kHz)	4.06
G.722 @ 56 kbit/s (wideband; 7kHz)	4.57

Considering the difficulties of evaluating signals having different bandwidths in the same test, these results are nevertheless able to indicate that there is a potentially significant benefit for the wideband solution over the narrowband case.

5.2.2 Wideband coding performance for clean speech (below 14.4 kbit/s)

At the start of the feasibility stage it was considered important to focus on solutions that can be realized using the 16 kbit/s A-ter interface. For this reason, the inputs to the study phase have mainly concentrated on solutions that are limited by 14.4 kbit/s, an estimate of the maximum speech data rate allowable in this application. The following table shows the results that have been presented for coding schemes working below the above mentioned limit.

Table V. Results for clean speech prototype wideband coding schemes ([9, 13])

	Nokia	DT/FT
16 kbit/s sub-multiplexing	3.9	4.46
G.722 48 kbit/s	3.9	4.53
G.722 56 kbit/s	4.4	4.72

This table shows that the quality of the ITU-T G.722 @ 48 kbit/s can be achieved by both of the coding schemes used. However, it can also be concluded that the quality of G.722 @ 56 kbit/s can not be realized by the codecs.

5.2.3 Wideband coding performance under background noise conditions (below 14.4 kbit/s)

Two experiments have been performed and presented that do consider the performance in background noise conditions ([7, 9]). However, the coding scheme used in the early presentation of Deutsche Telekom/France Telecom ([7]) is now outdated, since a new version of the coder has since delivered better results for clean speech ([13]), but was not tested in background noise. Therefore, only the Nokia results are presented here.

As a background noise signal, car noise at 15dB below the signal's energy was used in ([9]). The results are presented in Table VI.

Table VI. Results for speech+car noise using a prototype wideband coding scheme ([9])

	Nokia
16 kbit/s sub-multiplexing	4.2
G.722 48 kbit/s	4.3
G.722 56 kbit/s	4.7

Once again, it can be seen that the quality of G.722 @ 48 kbit/s can be reached using bit-rates below 14.4 kbit/s, while the G.722 @ 56 kbit/s poses difficulties.

It should be noted that G.722 at all bit-rates is very robust to certain kinds of background noise, since it tends to mask the quantization distortion in the presence of background noise.

5.2.4 Wideband coding performance under several channel conditions (below 14.4 kbit/s)

Two experiments have been presented to SMG11 describing the behavior of wideband coding scheme under different channel conditions ([9, 13]).

Table VII. Results for clean speech in different channel conditions using prototype wideband coding schemes ([9, 13])

	Nokia	DT / FT
G.722 @ 48 kbit/s	3.9	4.53
G.722 @ 56 kbit/s	4.4	4.72
no errors	4.4	4.46
16 dB C/I	N/A.	N/A.
13 dB C/I	3.5	4.46
10 dB C/I	3.2	3.69

7 dB C/I	1.7	2.97
4 dB C/I	1.2	N/A.

These results prove that it is feasible to realize a coding scheme that is robust to mild to moderate channel conditions. There is actually no degradation for the codec used in [13] down to 13 dB C/I (equiv. to G.722 @ 48 kbit/s) and a relatively graceful degradation down to 7 dB C/I.

5.2.5 Wideband coding performance under tandeming conditions (below 14.4 kbit/s)

The results presented in [9, 13] are summarized in Table VIII.

Table VIII. Results for speech in tandeming conditions using prototype wideband coding schemes ([9, 13])

	Nokia	**DT / FT**
16 kbit/s sub-multiplexing	3.3	3.78
G.722 48 kbit/s	3.1	4.25
G.722 56 kbit/s	4.2	4.66

Again, the solution presented in [9] shows that it is possible to achieve the quality of G.722 @ 48 kbit/s using bit-rates below 14.4 kbit/s.

5.2.6 Wideband coding at modes above 14.4 kbit/s

Although the main focus of the efforts made during the feasibility study was on coders complying with the 16 kbit/s-sub-multiplexing constraint, in [9], a wideband codec operating at a rate above 14.4 kbit/s and below 22.8 kbit/s was also tested. The results are summarized in Table IX.

Table IX. Results for different conditions using a higher bit-rate wideband codec.

	clean speech	**+car noise (SNR: 15dB)**	**tandeming (2)**
14.4 kbit/s< CuT < 22.8 kbit/s	4.2	4.6	3.8
G.722 48 kbit/s	3.9	4.3	3.1
G.722 56 kbit/s	4.4	4.7	4.2

This table indicates that G.722 @ 56 kbit/s is within reach when using bit-rates above 14.4 kbit/s and below 22.8 kbit/s. For good channel conditions, this quality can therefore be achieved within the GSM TCH-FS (full-rate) channel.

5.2.7 ITU-T standardization information relevant to AMR-WB feasibility study

It should be noted that the current requirements set by ITU-T for the WB coding scheme operating at 16 kbit/s asks for a proposal better than G.722 @ 48 kbit/s. Additionally, in the 24/32 kbit/s wideband coding competition, it was demonstrated that at 24 kbit/s it is possible to achieve equivalence to G.722 @ 56 kbit/s for speech in clean, background noise and tandeming conditions.

5.2.8 Summary

Based on the evidence presented in the last sections, it can be concluded that the initial performance objectives set for the feasibility study (i.e. equivalence to G.722 @ 56 kbit/s in clean and noisy speech) could be achieved with the GSM TCH-FS (full-rate) channel within the prospective time frame of the AMR-WB. In addition, the results show that an AMR-WB solution operating under the 16 kbit/s sub-multiplexing constraint will be capable of achieving equivalence to G.722 @ 48 kbit/s under sufficiently good channel conditions (down to 13 dB C/I).

5.3 Performance requirements specification

A draft working assumption for performance requirement specification is included in Annex B.

5.4 Design constraints

This section considers some features relevant as design constraints for the AMR-WB codec development. Some of these constraints are basically the same as in the AMR-NB standardization process. A draft working assumption for AMR-WB design constraints is given in Annex C. The design constraints need to be revised and finalized during the development phase.

6. Feasibility aspects for other channels

6.1 EDGE channels

It should be studied, during the development phase, if complementing performance requirements or design constraints need to be set in anticipation of use of AMR-WB in EDGE channels.

6.2 GSM multi-slot channels

This has not been studied during the feasibility phase but could be assessed in the development phase.

6.3 3G channels

This aspect has not been considered in detail. However, it can be foreseen that collaboration with the relevant 3G-groups (3GPP TSG S4) can and should be established on this issue.

7. Development phases

The target is to approve the AMR-WB codec by the end of year 2000 to be included in GSM release 2000.

The development will include the following main activities:

1. **Qualification phase:**
   - Finalize performance requirements and design constraints
   - Define Qualification Rules, Qualification Test Plan and List of Qualification Deliverables
   - Codec testing, result analysis and selection of codecs to proceed to selection
   - Results presentation to SMG for approval of AMR-WB to proceed into Selection Phase
2. **Selection Phase:**
   - Define Selection rules, Selection Test Plan and List of Selection Deliverables
   - Codec testing, result analysis and selection of codec
   - Results presentation to SMG for approval of AMR-WB codec
3. **Optimization phase (optional; to improve performance if needed)**
4. **Verification phase** (to verify that complexity and delay limits are met, and to provide additional information on codec performance)
5. **Preparation of specifications**

- Specifications preparation
- Characterization tests

The draft schedule for the AMR-WB codec development, testing, selection and standardization is shown in Annex E.

8. Open issues and risks

The main performance limitations and technical risk areas have been identified as follows:

- Codec performance in poor channel conditions: Sufficient error resilience in poor operating conditions must be verified.
- Architectural and network issues
- Acoustical design of terminals
- Call setup
- Signaling
- Transmission

9. Conclusions

The AMR-WB feasibility study has been ongoing since October 1997. A number of organizations have contributed with performance results, as well as with proposals for requirements and constraints.

The feasibility study has concentrated on the possibility to offer a high quality wideband service using the GSM full-rate traffic channel. The studies performed indicate that it is feasible to develop an AMR-WB service operating on the GSM full-rate traffic channel (22.8 kbit/s) with a quality for speech approaching that of G.722 at 56 kbit/s for good radio conditions. This is in line with the performance objective from the early discussions on AMR-WB. If GSM 16 kbit/s A-ter sub-multiplexing is used, thus further limiting the highest source coder rate, it is still likely that the quality for speech will be as good as G.722 at 48 kbit/s.

SMG11 believes that the AMR-WB coder should be a multi-rate wideband speech coder. It is further believed that channel quality based mode adaptation similar to the AMR-NB is required to achieve the performance goals.

At a relatively late stage, applications using higher bit-rate mobile channels were added to the study. GSM/EDGE phase II as well as the 3GPP systems are now included in the scope of the activity. Using higher rate channels, it is possible to make use of AMR-WB modes with rates exceeding 22.8 kbit/s, thus achieving an even higher level of quality.

In light of the feasibility study results, SMG11 believes that an AMR-WB project is feasible. The draft working assumption for the performance requirements present a challenging yet realistic set of quality goals which will enable high quality wideband speech services.

10. Recommendations

On basis of the performance benefits and the risks highlighted above, the following recommendations are submitted to SMG for approval:

- Initiate a program to develop, test, select and specify AMR wideband (AMR-WB) together with related features such as VAD/DTX.
- An acceptable and workable timeplan should be agreed by SMG11 (and SMG1/SMG2/SMG3/SMG12) as soon as possible after SMG#29 with the priority of targeting approval of AMR-WB codec by end of year 2000 for GSM Release 2000.
- Approve Work Item description in Annex F

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Annex A. Terminology

Annex B. Draft working assumption for performance specification

Annex C. Draft working assumption for design constraints

Annex D. Implementation requirements (draft)

Annex E. Draft schedule of AMR-WB development

Annex F. Work item description for AMR Wideband

Annex A. Terminology

The terminology used in this document and recommended for other work on AMR is listed below.

Adaptive Multi-rate (AMR) codec	Speech and channel codec capable of operating at various combinations of speech and channel coding (*codec mode*) bit-rates
Channel mode	Half-rate or full-rate operation
Channel mode adaptation	The control and selection of the *channel radio bearer capability (e.g. in TCH-FS and EDGE half-rate or full-rate operation).*
Codec mode	For a given *channel mode*, the bit partitioning between the speech and channel codecs.
Codec mode adaptation	The control and selection of the *codec mode* bit-rates. Normally, implies no change to the *channel mode*.
Full-rate (FR)	Full-rate channel or *channel mode*
Gross bit-rate	The bit-rate of the *channel mode* selected.
Half-rate (HR)	Half-rate channel or *channel mode*
In-Band Signaling	Signaling for DTX, Link Control, Channel and codec mode modification, etc. carried within the traffic channel by reserving or stealing bits normally used for speech transmission. Maybe on the radio channel or other channels inside the fixed network (e.g. A-bis, A-ter, A).
Out-of-Band Signaling	Signaling on the GSM control channels to support link control. May be on the radio channel of other channels inside the fixed network (e.g. A-bis, A-ter, and A). Note: Out-Of-Band Signaling on the radio channel sometimes “steals” capacity from the speech traffic channel (FACCH) thus creating speech distortion.
Toll Quality	Speech quality normally achieved on modern wireline telephones. Synonym with "ISDN quality".
Wireline quality	Speech quality provided by modern wireline networks. Normally taken to imply quality at least as good as that of 32 kbit/s G.726 or 16 kbit/s G.728 codecs.

Acronyms

AMR	Adaptive Multi-rate
BER	Bit Error Rate
C/I	Carrier-to-Interferer ratio
CuT	Codec under Test
DSP	Digital Signal Processor
DTX	Discontinuous Transmission for power consumption and interference reduction
EDGE	Enhanced Data for GSM Evolution
EFR	Enhanced Full-rate
ETSI	European Telecommunications Standards Institute
FR	Full-rate
FH	Frequency Hopping
G.726	ITU 16/24/32/40 kbit/s ADPCM codec
G.728	ITU 16 kbit/s LD-CELP codec
G.722	ITU 48/56/64 kbit/s subband ADPCM wideband codec
G.711	ITU 64 kbit/s PCM codec
GSM	Global System for Mobile communications
HR	Half-rate
ITU-T	International Telecommunication Union - Telecommunications Standardization Sector
MNRU	Modulated Noise Reference Unit
MOS	Mean Opinion Score
MS	Mobile Station
NB	Narrowband
AMR-NB	AMR Narrowband
PCM	Pulse Code Modulation
SMG	Special Mobile Group
SNR	Signal to Noise Ratio
SG16	Study Group 16
TCH-HS	Traffic CHannel Half-rate Speech
TCH-FS	Traffic CHannel Full-rate Speech
TFO	Tandem Free Operation
VAD	Voice Activity Detector
WB	Wideband
AMR-WB	AMR Wideband
3GPP	3rd Generation Partnership Project

Annex B. Draft working assumption for performance specification

Introduction

This document contains draft working assumptions for 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. General (this section may be moved from this document)

The design constraints and the performance requirements for the AMR WB coder are set to provide a wideband coder, which is applicable to GSM and evolved or future mobile systems. The similarities in terms of requirements for the different systems are substantial; e.g. the required error robustness, the most important usage scenarios and the approximate bit-rate range.

For GSM, 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 GSM/EDGE phase II channels

D GSM multi-slot traffic channels (n*22.8 kbit/s)

The AMR WB coder is a multi-rate coder. It is required that the same basic coder is used for all applications. For the development, the focus will be on cases A and B, which does not exclude coders with source coding modes higher than 22.8 kbit/s. It is expected that the AMR WB coder will be a channel quality adaptive coder in a similar manner to the AMR NB coder.

2. 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.

C/I	Application	Performance requirement	Performance objective
no errors	A	G.722-48k	G.722-56k
	B	G.722-56k	G.722-64k
19 dB	A	G.722-48k	
	B	G.722-56k	
16 dB	A	G.722-48k	
	B	G.722-48k	
13 dB	A	G.722-48k	
	B	G.722-48k	
10 dB	A	t.b.d.	
	B	t.b.d	
7 dB	A	t.b.d	
	B	t.b.d	
4 dB	A	t.b.d	
	B	t.b.d	

Table 1a: Clean speech requirements under static test conditions.

C/I	Application	Performance requirement	Performance objective
no errors	A	G.722-48k	G.722-56k
	B	G.722-56k	G.722-64k
19 dB	A	G.722-48k	
	B	G.722-48k	
16 dB	A	G.722-48k	
	B	G.722-48k	
13 dB	A	G.722-48k	
	B	G.722-48k	
10 dB	A	t.b.d.	
	B	t.b.d	
7 dB	A	t.b.d	
	B	t.b.d	
4 dB	A	t.b.d	
	B	t.b.d	

Table 1b: Background noise requirements under static test conditions:

For AMR WB bit-rates exceeding 22.8 kbit/s, the requirement for speech and speech in background noise is G.722 at 64 kbit/s.

3. 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	Same or better than the EFR under the same conditions

Table 2: Requirements under dynamic test conditions

4. Additional speech codec performance requirements and objectives

The reference speech codecs for the performance under tandeming and talker, level and language dependency are specified in Table 3. The performance requirements and objectives for DTMF, information tones and idle noise are specified in Table 4.

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 characterization phase.

Condition	**Reference for highest bit-rate** **Requirement**	**Objective**
Tandeming	t.b.d	
Talker dependency	t.b.d.	
Level dependency	t.b.d.	
Language dependency	t.b.d.	

Table 3: Reference codecs for additional speech signal performance requirements.

Condition	**Requirement**	**Objective**
DTMF		*Transparent transmission of DTMF.*
Information tones	*Recognizable as given information tone.*	

Table 4: Requirements and objectives for speech codec performance with non-speech inputs

5. Open Issues

This section lists open issues currently under discussion.

- Performance for non-speech signals such as:

  - Music requirements for higher rate AMR WB modes?

  - DTMF (applicability to be clarified) capability

  - Information (call progress) tones

- Performance in tandem
- Requirement specification method for 10, 7, and 4 dB C/I

Annex C. Draft working assumption for design constraints

The following table provides draft working assumptions for design constraints. The design constraints should be reviewed and finalized during the development phase.

Development constraints	**Open issues, notes**
Complexity requirements: The complexity requirements are valid for all phases of the AMR-WB development and are separate for channel coding, speech coding and DTX algorithms. *Channel coding including possible control loop management algorithms:* **A. wMOPS(AMR-WB ch. codec)** ≤ **5.2 wMOPS** ≈ wMOPS(AMR-NB FR ch. codec: 5.15) **B. RAM(AMR-WB ch. Codec)** ≤ **2.5 kwords** ≈ RAM(AMR-NB FR ch. codec: 2.427 kwords) **C. ROM(AMR-WB ch. codec)** ≤ **2.8 kwords** ≈ ROM(AMR-NB ch. codec: 2.739 kwords) **D. Program ROM(AMR-WB ch. Codec.)** ≤ **Program ROM(AMR-NB FR ch. Codec)** ≈ ROM(AMR-NB FR ch. codec: 1 366 ETSI basic operators) *Speech coding (excluding VAD/DTX):* **E. wMOPS** ≤ **40 wMOPS** ≈ 2.4 x wMOPS(AMR-NB sp. codec: 16.61) **F. RAM(AMR-WB speech codec)** ≤ **15 kwords** ≈ 2.6 × RAM(AMR-NB speech codec: 5.819 kwords) **G. ROM(AMR-WB speech codec)** ≤ **15 kwords** ≈ 1.0 x ROM(AMR-NB speech codec 14.343 kwords) **H. Program ROM(AMR-WB sp. Codec)** ≤ **Program ROM(AMR-NB speech codec)** ≈ ROM(AMR-NB speech codec: 4 830 ETSI basic operators) *Additional complexity for VAD/DTX operation:* *t.b.d.* Notes: • Program ROM is computed as the number of basic instructions • 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 signaling. **Complexity calculation rules:** The same complexity evaluation methodology as used in the past for GSM AMR narrowband standardization (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**: 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). • **Selection**: ETSI methodology based on fixed point code (Basic op. Counters, i.e. Worst observed case) • **Verification/characterization**: ETSI methodology based on fixed point code (Theoretical worst case)	WB = Wideband NB = Narrowband

Arithmetic used in codec proposals: • **Qualification**: Fixed point or floating point code • **Selection**: Fixed point code (using ETSI set of basic operations) • **Verification/characterization**: Fixed-point code (using ETSI set of basic operations).	
A-ter sub-multiplexing / constraints for bit-rates At least one codec mode at AMR-WB shall be consistent with 16 kbit/s sub-multiplexing 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 t.b.d.]. All source codecs below or equal to 22.8 kbit/s must contain channel coding to the single slot gross bit-rate (22.8 kbit/s). The modes below [14.4 kbit/s t.b.d.] can be used within 16 kbit/s sub-multiplexing. (The performance requirements for the modes and their testing are explained in AMR-WB Performance Requirements document and in Qualification and Selection Test Plan documents.)	Maximum bit-rate for 16 kbit/s sub-multiplexing must be defined
Channel mode: The AMR-WB codec will operate only in Full-rate speech traffic channel (TCH-FS). It is possible to make a channel mode handovers between AMR-WB FR and AMR-NB HR channels in the same way as existing intra-cell handovers. This 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 coding: The existing sets of convolutional polynomials used in the GSM shall be used.	
In-band signaling and codec mode control: AMR-WB may use the existing in-band AMR signaling, additional signaling to the existing in-band AMR signaling, or a new signaling. The channel quality measurement and mode adaptation may also be different than in existing AMR. Similarities to existing AMR should be preferred for compatibility and implementation reasons unless performance improvement can be obtained otherwise.	
Tandem Free Operation (TFO): The AMR-WB codec shall support Tandem Free Operation TFO mode can be operated only if both terminals (e.g. up-link MS-A to network and downlink, network to MS-B) use the same speech coding bit-rate and algorithm.	
Discontinuous Transmission (DTX): Similarities to existing AMR should be preferred unless performance improvement can be obtained otherwise. VAD/DTX is not developed until later stage of the codec development. The proponents of the selected codec shall provide the VAD/DTX solution.	
Active noise suppression in the selection 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 standardization of a noise suppresser may then be addressed in a separate phase	
Transmission delay: The target is to keep the round trip delay for wideband modes equal to the round trip delay of the GSM AMR FR. Nevertheless, some increase of transmission delay is expected due to the higher source coding bit-rates in AMR-WB. A suitable limit for transmission delay shall be set during the AMR-WB development phase.	The possibly allowed additional delay must be defined.
Error concealment: Error concealment techniques of AMR-WB codec candidates shall only rely on soft-output information from the equalizer (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 audio bandwidth will be [50 Hz] to 7 kHz	

Annex D. Implementation requirements (draft)

The following basic functions are impacted by the introduction of AMR wideband mode(s):

Device	**Upgrade for AMR Wideband**
MS	New source codec New channel codec New audio parts: microphone, earpiece, A/D-converter, D/A-converter, audio bandwidth filters, acoustic design
BTS	New channel codec New TRAU-frames (possibly also for higher than 16 kbit/s bit-rate sub-multiplexing)
BSC	t.b.d.
TRAU	New TRAU-frames (possibly also for higher than 16 kbit/s bit-rate sub-multiplexing) New source codec
MSC	t.b.d.
OMC-P	t.b.d.

Annex E. Draft schedule of AMR-WB development

The draft schedule for AMR-WB development and standardization is outlined below.

Month	**Meeting / date**	**Activity**
June 1999	SMG#29 (23-25 June)	Launch of AMR-WB standardization
July 1999		Start work of preparing Test Plan for AMR-WB Qualification (by correspondence)
August 1999		
September 1999		
October 1999	SMG11#12 (4-8 October)	SMG11 approval of • Design Constraints • Performance Requirements Indication of number of candidates (decision of need for a separate Qualification Phase) Work on Test Plans, Qualification rules and Qualification Deliverables
November 1999	SMG#30 (8-12 November)	Presentation of Design Constraints and Performance Requirements for SMG approval
December 1999		
January 2000	SMG11#13 (late January)	SMG11 approval of • Qualification Test Plan • Qualification and selection Rules • List of Qualification Deliverables
February 2000	SMG#31 (14-18 February)	Qualification tests (in house including some cross testing) Preparation of Qualification Deliverables
March 2000		
April 2000		
May 2000	SMG11#14	Submission of: • Qualification Test Results • Qualification Deliverables SMG11 approval of • List of Selection Deliverables • Selection Test plan • Codecs to proceed into selection
June 2000	SMG#32 (13-16 June)	Presentation of Qualification Test results for approval

July 2000		Submission of AMR-WB candidates for Selection Tests Selection testing: host lab processing, listening tests, and results analysis Preparation of Selection Deliverables
August 2000		
September 2000		
October 2000	SMG11#15	Submission of: • Selection Deliverables Selection of AMR-WB codec
November 2000	SMG#33 (6-10 November)	SMG approval of AMR-WB codec
December 2000		
January 2001		
February 2001	SMG#34?	SMG approval of all AMR-WB specifications

Annex F. Work item description for AMR wideband (AMR-WB)

1. AMR-Wideband codec

1.1SMG Work Area

	UMTS Radio Access
X	GSM Radio Access
	GSM-UMTS Core Network
	UMTS Services

1.2Linked work items

Adaptive Multi-rate codec

1.3Justification

While current GSM codecs achieve good performance for narrowband speech (audio bandwidth limited below 3.4 kHz), the introduction of a wideband speech service (audio bandwidth extended to 7 kHz) would provide improved voice quality especially in terms of increased naturalness in voice. Wideband coding would bring speech quality exceeding that of (narrowband) wireline quality to GSM. There is a growing market interest in a wideband speech service among operators and in having a wideband codec included into GSM AMR as a new optional wideband mode (or modes).

1.4Service Aspects

The main requirement for the AMR-WB codec is to provide wideband (7 kHz audio bandwidth) coding of speech exceeding narrowband speech quality.

1.5MMI-Aspects

None

1.6Charging Aspects

???

1.7Security Aspects

None

1.8 Impacts

Affects:	USIM	ME	NW	Others
Yes		x	x	
No	x			
Don't know				

1.9Expected Output and Timescales (to be updated at each plenary)

New specifications						
Spec No.	Title	Prime rsp. STC	2ndary rsp. STC(s)	presented for information at SMG#	approved at SMG#	Comments
		SMG11	SMG1 SMG2 SMG3 SMG12			

Affected existing specifications				
Spec No.	CR	Subject	Approved at SMG#	Comments

1.10 Work item rapporteurs

Imre Varga, Siemens

1.11 Work item leadership

SMG11

1.12 Supporting companies

Deutsche Telekom

Ericsson

France Telecom

Mannesmann

Nokia

Siemens

Texas Instruments

1.13 Others