**3GPP TSG SA4 meeting #25bis S4-030248**

**Berlin, Germany, 24. –28.Feb 2003 Agenda Item: SES**

**Title: Design Constraints for default codec for speech enabled services (SES)**

# Source: SQ-SWG

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**Version: 1.4**

**Summary:**

This document describes the design constraints for the codecs for speech enabled services

# Sampling Rates

Sampling rates of 8 & 16kHz will be supported.

# Complexity

The terminal side processing of the codec has to be able to be implemented within the resources of a typical mobile phone terminal. Accordingly the maximum complexity requirements for terminal side codec have been defined as shown in tables below. Table 1 shows complexity requirements for codec supporting 8kHz sampling rate and table 2 shows numbers for codec supporting 16 kHz sampling rate.

|  |  |
| --- | --- |
| **Measure** | **Requirement** |
| WMOPS | Less than 25 |
| ROM size | Less than 20 kwords |
| RAM size | Less than 7 kwords |

Table 1: complexity and memory requirements for codec supporting 8 kHz sampling rate

|  |  |
| --- | --- |
| **Measure** | **Requirement** |
| WMOPS | Less than 39 |
| ROM size | Less than 34 kwords |
| RAM size | Less than 8 kwords |

**Table 2: complexity and memory requirements for codec supporting 16 kHz sampling rate** The definition of the wMOPS measure and recommendations on how to estimate the computation and memory requirements can be found in ETSI Technical document [2]. A word is defined as 16bits. These complexity measures are for the front-end feature extraction and compression and the VAD.

ROM does not include program ROM.

# Latency

The maximum codec latency requirement is 200ms, with the objective of 50 ms. This values contains the algorthmic delay introduced by the codec.

# Data rate for the source codec

#### Voice enabled services need to be able to operate over a variety of channels. The following channels and datarates will at least be supported

#### a) For conversational class of service [4]:

* The GPRS single slot uplink (Coding scheme CS-1) channel.   
  Here the maximum source data rate is 5.6 kbit/sec.
* The EGPRS single slot uplink (Coding scheme MCS -1) channel.  
  Here the maximum source data rate is 6.4 kbit/sec.
* The Flexible Layer 1 (FLO) channel. Here the maximum data rate is expected to be between 6.4 and 8.4 kbit/sec.
* For UTRAN packet data channel the maximum source datarate is 24 kbit/sec.

It is assumed one 20ms frame within one RLC/MAC block.

b) For streaming and interactive class of service\_

* For GPRS / EGPRS single slot uplink channel the maximum source datarate is 8 kbit/sec (assuming 10 frames per IP packet) or 7.5 kbit/sec (assuming 5 frames per IP packet) .
* For UTRAN packet data channel the maximum source datarate is 24 kbit/sec.

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1. ETSI SMG11 Tdoc SMG11 117/99, “Complexity verification report of the AMR codec, v2.0”, Alcatel, Philips, ST Microelectronics, Texas Instruments".
2. ETSI SMG11 AMR-9 "AMR permanent document (AMR-9) Complexity and delay assessment v1.0", 23rd March 1998
3. IETF RFC 3095: "RObust Header Compression (ROHC): Framework and four profiles: RTP, UDP, ESP, and uncompressed".
4. S4-030114.doc , TSG SA WG4 , Berlin, Germany, 24. –28.Feb 2003