

ETSI TS 126 194 V10.0.1 (2012-03)



**Digital cellular telecommunications system (Phase 2+);
Universal Mobile Telecommunications System (UMTS);
Speech codec speech processing functions;
Adaptive Multi-Rate - Wideband (AMR-WB) speech codec;
Voice Activity Detector (VAD)
(3GPP TS 26.194 version 10.0.1 Release 10)**



Reference

RTS/TSGS-0426194va01

Keywords

GSM,UMTS

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:

<http://www.etsi.org>

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at

<http://portal.etsi.org/tb/status/status.asp>

If you find errors in the present document, please send your comment to one of the following services:

http://portal.etsi.org/chaicor/ETSI_support.asp

Copyright Notification

No part may be reproduced except as authorized by written permission.
The copyright and the foregoing restriction extend to reproduction in all media.

© European Telecommunications Standards Institute 2012.
All rights reserved.

DECT™, **PLUGTESTS™**, **UMTS™** and the ETSI logo are Trade Marks of ETSI registered for the benefit of its Members.
3GPP™ and **LTE™** are Trade Marks of ETSI registered for the benefit of its Members and of the 3GPP Organizational Partners.
GSM® and the GSM logo are Trade Marks registered and owned by the GSM Association.

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (<http://ipr.etsi.org>).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This Technical Specification (TS) has been produced by ETSI 3rd Generation Partnership Project (3GPP).

The present document may refer to technical specifications or reports using their 3GPP identities, UMTS identities or GSM identities. These should be interpreted as being references to the corresponding ETSI deliverables.

The cross reference between GSM, UMTS, 3GPP and ETSI identities can be found under <http://webapp.etsi.org/key/queryform.asp>.

Contents

Intellectual Property Rights	2
Foreword.....	2
Foreword.....	4
1 Scope	5
2 Normative References	5
3 Technical Description.....	5
3.1 Definitions, symbols and abbreviations.....	5
3.1.1 Definitions	5
3.1.2 Symbols	5
3.1.2.1 Variables	5
3.1.2.2 Constants	6
3.1.2.3 Functions	7
3.1.3 Abbreviations.....	8
3.2 General	8
3.3 Functional description	8
3.3.1 Filter bank and computation of sub-band levels	9
3.3.2 Tone detection	10
3.3.3 VAD decision	11
3.3.3.1 Hangover addition.....	12
3.3.3.2 Background noise estimation	13
3.3.3.3 Speech level estimation.....	14
4 Computational details.....	14
Annex A (informative) : Change history	15
History	16

Foreword

This Technical Specification has been produced by the 3GPP.

This document specifies the Voice Activity Detector (VAD) to be used in the Discontinuous Transmission (DTX) as described in [3].

The contents of the present document are subject to continuing work within the TSG and may change following formal TSG approval. Should the TSG modify the contents of this TS, it will be re-released by the TSG with an identifying change of release date and an increase in version number as follows:

Version x.y.z

where:

- x the first digit:
 - 1 presented to TSG for information;
 - 2 presented to TSG for approval;
 - 3 Indicates TSG approved document under change control.
- 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;

1 Scope

This document specifies the Voice Activity Detector (**VAD**) to be used in the Discontinuous Transmission (**DTX**) as described in [3].

The requirements are mandatory on any **VAD** to be used either in User Equipment (**UE**) or Base Station Systems (**BSS**)s that utilize the **AMR** wideband speech codec.

2 Normative References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication, edition number, version number, etc.) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

- [1] 3GPP TS 26.173: "ANSI-C code for the Adaptive Multi-Rate Wideband speech codec" .
- [2] 3GPP TS 26.190: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions" .
- [3] 3GPP TS 26.193: "Speech codec speech processing functions; Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Source controlled rate operation".
- [4] ITU, The International Telecommunications Union, Blue Book, Vol. III, Telephone Transmission Quality, IXth Plenary Assembly, Melbourne, 14-25 November, 1988, Recommendation G.711, Pulse code modulation (PCM) of voice frequencies.
- [5] 3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

3 Technical Description

3.1 Definitions, symbols and abbreviations

3.1.1 Definitions

For the purposes of the present document, the terms and definitions given in TR 21.905 [5] and the following apply. A term defined in the present document takes precedence over the definition of the same term, if any, in TR 21.905 [5].

frame: Time interval of 20 ms corresponding to the time segmentation of the speech transcoder.

3.1.2 Symbols

For the purposes of this TS, the following symbols apply.

3.1.2.1 Variables

bckr_est[n] background noise estimate at the frequency band "n"

burst_count	counts length of a speech burst, used by VAD hangover addition
hang_count	hangover counter, used by VAD hangover addition
level[n]	signal level at the frequency band "n"
new_speech	pointer of the speech encoder, points a buffer containing last received samples of a speech frame [2]
noise_level	estimated noise level
pow_sum	input power
s(i)	samples of the input frame
snr_sum	measure between input frame and noise estimate
speech_level	estimated speech level
stat_count	stationary counter
stat_rat	measure indicating stationary of the input frame
tone_flag	flag indicating the presence of a tone
vad_thr	VAD threshold
VAD_flag	Boolean VAD flag
vadreg	intermediate VAD decision

3.1.2.2 Constants

ALPHA_UP1	constant for updating noise estimate (see subclause 3.3.5.2)
ALPHA_DOWN1	constant for updating noise estimate (see subclause 3.3.5.2)
ALPHA_UP2	constant for updating noise estimate (see subclause 3.3.5.2)
ALPHA_DOWN2	constant for updating noise estimate (see subclause 3.3.5.2)
ALPHA3	constant for updating noise estimate (see subclause 3.3.5.2)
ALPHA4	constant for updating average signal level (see subclause 3.3.5.2)
ALPHA5	constant for updating average signal level (see subclause 3.3.5.2)
BURST_HIGH	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
BURST_P1	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
BURST_SLOPE	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
COEFF3	coefficient for the filter bank (see subclause 3.3.1)
COEFF5_1	coefficient for the filter bank (see subclause 3.3.1)
COEFF5_2	coefficient for the filter bank (see subclause 3.3.1)
HANG_HIGH	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
HANG_LOW	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
HANG_P1	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
HANG_SLOPE	constant for controlling VAD hangover addition (see subclause 3.3.5.1)
FRAME_LEN	size of a speech frame, 256 samples (20 ms)

MIN_SPEECH_LEVEL1	constant for speech estimation (see subclause 3.3.5.3)
MIN_SPEECH_LEVEL2	constant for speech estimation (see subclause 3.3.5.3)
MIN_SPEECH_SNR	constant for VAD threshold adaptation (see subclause 3.3.5)
NO_P1	constant for VAD threshold adaptation (see subclause 3.3.5)
NO_SLOPE	constant for VAD threshold adaptation (see subclause 3.3.5)
NOISE_MAX	maximum value for noise estimate (see subclause 3.3.5.2)
NOISE_MIN	minimum value for noise estimate (see subclause 3.3.5.2)
POW_TONE_THR	threshold for tone detection (see subclause 3.3.5)
SP_ACTIVITY_COUNT	constant for speech estimation (see subclause 3.3.5.3)
SP_ALPHA_DOWN	constant for speech estimation (see subclause 3.3.5.3)
SP_ALPHA_UP	constant for speech estimation (see subclause 3.3.5.3)
SP_CH_MAX	constant for VAD threshold adaptation (see subclause 3.3.5)
SP_CH_MIN	constant for VAD threshold adaptation (see subclause 3.3.5)
SP_EST_COUNT	constant for speech estimation (see subclause 3.3.5.3)
SP_P1	constant for VAD threshold adaptation (see subclause 3.3.5)
SP_SLOPE	constant for VAD threshold adaptation (see subclause 3.3.5)
STAT_COUNT	threshold for stationary detection (see subclause 3.3.5.2)
STAT_THR	threshold for stationary detection (see subclause 3.3.5.2)
STAT_THR_LEVEL	threshold for stationary detection (see subclause 3.3.5.2)
THR_HIGH	constant for VAD threshold adaptation (see subclause 3.3.5)
TONE_THR	threshold for tone detection (see subclause 3.3.3)
VAD_POW_LOW	constant for controlling VAD hangover addition (see subclause 3.3.5.1)

3.1.2.3 Functions

+	Addition
-	Subtraction
*	Multiplication
/	Division
x	absolute value of x
AND	Boolean AND
OR	Boolean OR

$$\sum_{n=a}^b x(n) = x(a) + x(a+1) + \dots + x(b-1) + x(b)$$

$$\text{MIN}(x,y) = \begin{cases} x, & x \leq y \\ y, & y < x \end{cases}$$

$$\text{MAX}(x,y) = \begin{cases} x, & x \geq y \\ y, & y > x \end{cases}$$

3.1.3 Abbreviations

For the purposes of the present document, the abbreviations given in TR 21.905 [5] and the following apply. An abbreviation defined in the present document takes precedence over the definition of the same abbreviation, if any, in TR 21.905 [5].

ANSI	American National Standards Institute
DTX	Discontinuous Transmission
VAD	Voice Activity Detector
CNG	Comfort Noise Generation

3.2 General

The function of the VAD algorithm is to indicate whether each 20 ms frame contains signals that should be transmitted, e.g. speech, music or information tones. The output of the VAD algorithm is a Boolean flag (VAD_flag) indicating presence of such signals.

3.3 Functional description

The block diagram of the VAD algorithm is depicted in Figure 1. The VAD algorithm uses parameters of the speech encoder to compute the Boolean VAD flag (VAD_flag). This input frame for VAD is sampled at the 6.4 kHz frequency and thus it contains 256 samples. Samples of the input frame (s(i)) are divided into sub-bands and level of the signal (level[n]) in each band is calculated. Input for the tone detection function are the normalized open-loop pitch gains which are calculated by open-loop pitch analysis of the speech encoder. The tone detection function computes a flag (tone_flag) which indicates presence of a signalling tone, voiced speech, or other strongly periodic signal. Background noise level (bckr_est[n]) is estimated in each band based on the VAD decision, signal stationarity and the tone-flag. Intermediate VAD decision is calculated by comparing input SNR (level[n]/bckr_est[n]) to an adaptive threshold. The threshold is adapted based on noise and long term speech estimates. Finally, the VAD flag is calculated by adding hangover to the intermediate VAD decision.

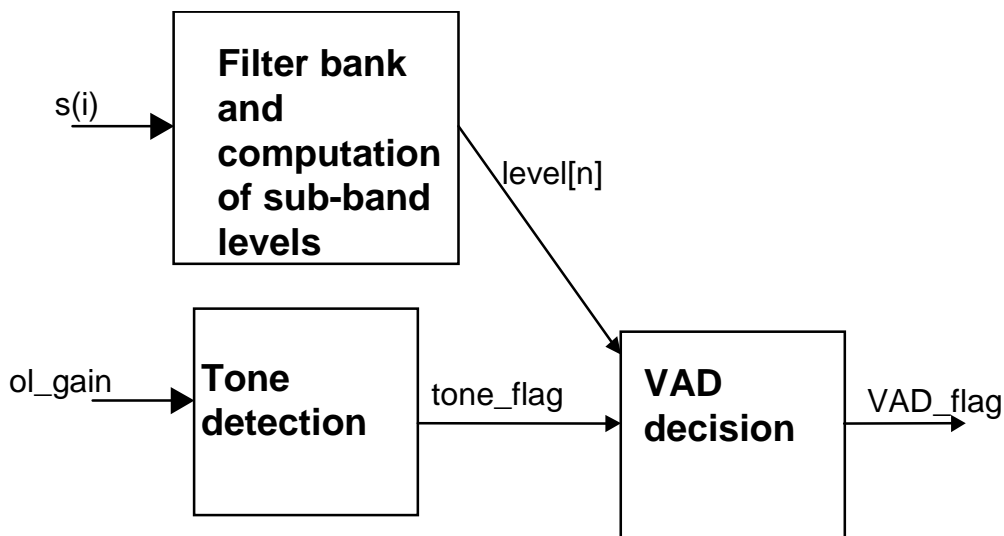


Figure 1: Simplified block diagram of the VAD algorithm

3.3.1 Filter bank and computation of sub-band levels

The input signal is divided into frequency bands using a 12-band filter bank (Figure 2). Cut-off frequencies for the filter bank are shown in Table 1.

Table 1. Cut-off frequencies for the filter bank

Band number	Frequencies
1	0 – 200 Hz
2	200 – 400 Hz
3	400 – 600 Hz
4	600 – 800 Hz
5	800 – 1200 Hz
6	1200 – 1600 Hz
7	1600 – 2000 Hz
8	2000 – 2400 Hz
9	2400 - 3200 Hz
10	3200 – 4000 Hz
11	4000 – 4800 Hz
12	4800 – 6400 Hz

Input for the filter bank is a speech frame pointed by the new_speech pointer of the speech encoder [1]. Input values for the filter bank are scaled down by one bit. This ensures safe scaling, i.e. saturation can not occur during calculation of the filter bank.

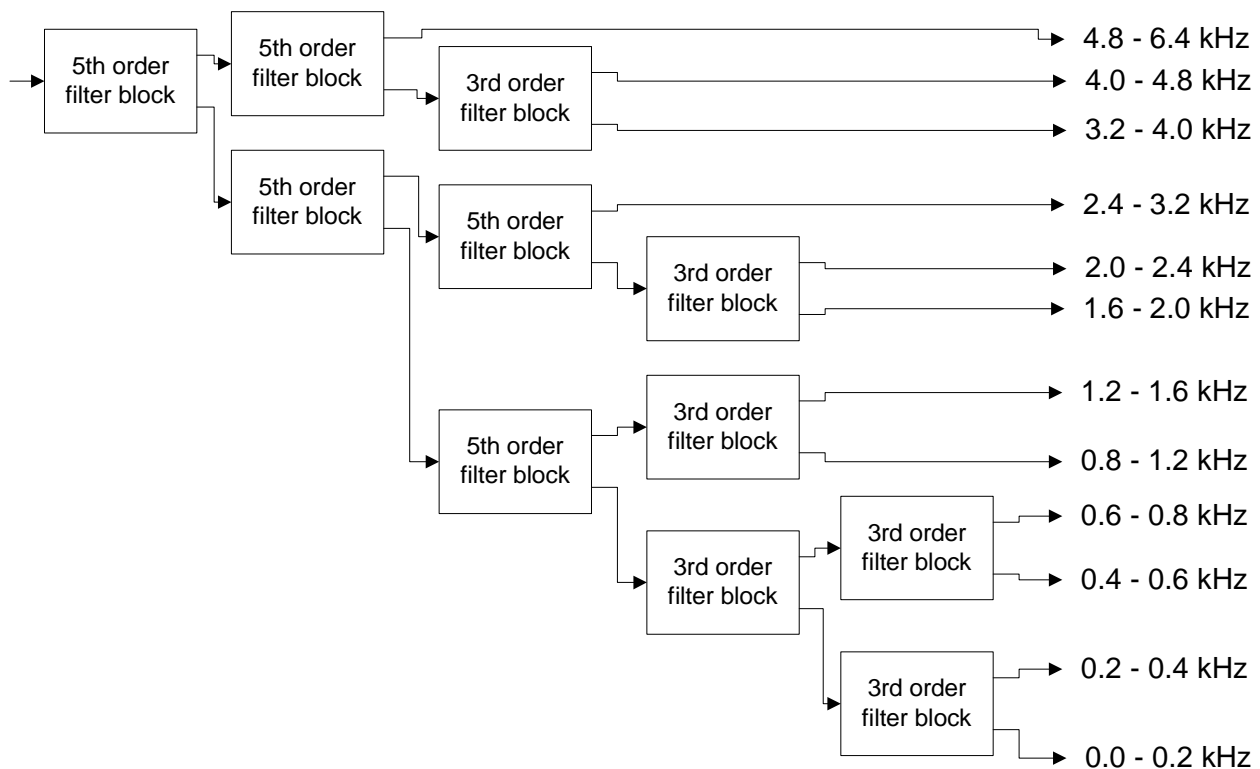


Figure 2: Filter bank

The filter bank consists of 5th and 3rd order filter blocks. Each filter block divides the input into high-pass and low-pass parts and decimates the sampling frequency by 2. The 5th order filter block is calculated as follows:

$$x_{lp}(i) = 0.5 * (A_1(x(2 * i)) + A_2(x(2 * i + 1))) \tag{1a}$$

$$x_{hp}(i) = 0.5 * (A_1(x(2*i)) - A_2(x(2*i+1))) \quad (1b)$$

where

$x(i)$ input signal for a filter block

$x_{lp}(i)$ low-pass component

$x_{hp}(i)$ high-pass component

The 3rd order filter block is calculated as follows:

$$x_{lp}(i) = 0.5 * (x(2*i+1) + A_3(x(2*i))) \quad (2a)$$

$$x_{hp}(i) = 0.5 * (x(2*i+1) - A_3(x(2*i))) \quad (2b)$$

The filters $A_1()$, $A_2()$, and $A_3()$ are first order direct form all-pass filters, whose transfer function is given by:

$$A(z) = \frac{C + z^{-1}}{1 + C * z^{-1}}, \quad (3)$$

where C is the filter coefficient.

Coefficients for the all-pass filters $A_1()$, $A_2()$, and $A_3()$ are COEFF5_1, COEFF5_2, and COEFF3, respectively.

Signal level is calculated at the output of the filter bank at each frequency band as follows:

$$level(n) = \sum_{i=START_n}^{END_n} |x_n(i)|, \quad (4)$$

where:

n index for the frequency band

$x_n(i)$ sample i at the output of the filter bank at frequency band n

$$START_n = \begin{cases} -6, & 1 \leq n \leq 4 \\ -12, & 5 \leq n \leq 8 \\ -24, & 9 \leq n \leq 11 \\ -48, & n = 12 \end{cases}$$

$$END_n = \begin{cases} 7, & 1 \leq n \leq 4 \\ 15, & 5 \leq n \leq 8 \\ 31, & 9 \leq n \leq 11 \\ 63, & n = 12 \end{cases}$$

Negative indices of $x_n(i)$ refer to the previous frame.

3.3.2 Tone detection

The purpose of the tone detection function is to detect information tones, vowel sounds and other periodic signals. The tone detection uses normalized open-loop pitch gains (ol_gain), which are received from the speech encoder. If the pitch gain is higher than the constant TONE_THR, tone is detected and the tone flag is set:

if (ol_gain > TONE_THR)

tone_flag = 1

The open-loop pitch search and correspondingly the tone flag is computed twice in each frame, except for mode 6.60 kbit/s, where it is computed only once.

3.3.3 VAD decision

The block diagram of the VAD decision algorithm is shown in figure 3.

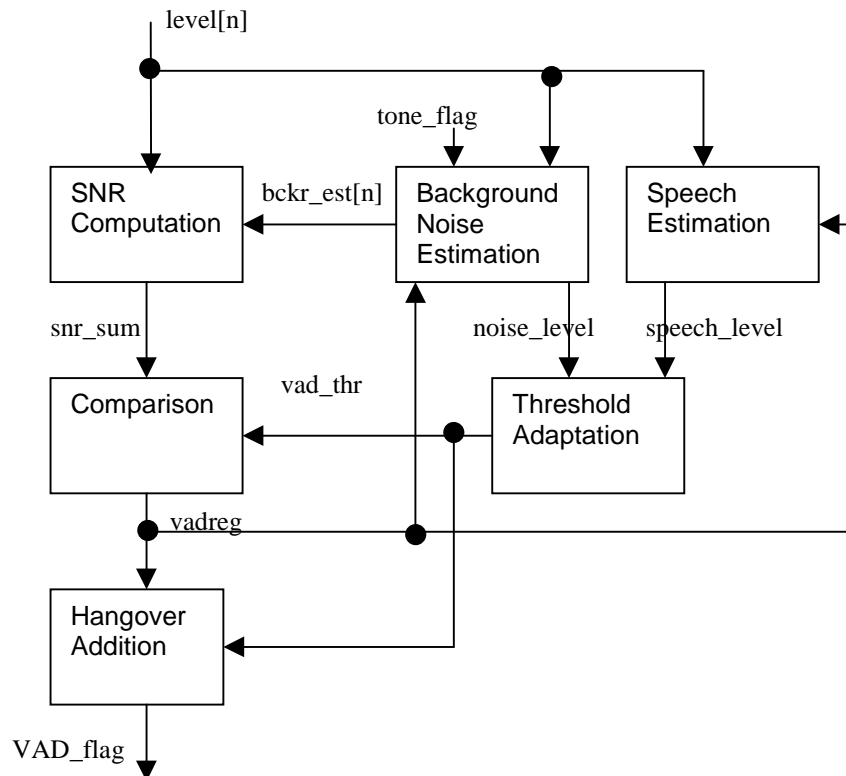


Figure 3: Simplified block diagram of the VAD decision algorithm

Power of the input frame is calculated as follows:

$$frame_pow = \sum_{i=0}^{FRAME_LEN} s(i) * s(i), \tag{5}$$

where samples s(i) of the input frame are pointed by the new_speech pointer of the speech encoder. Variable pow_sum is sum of the powers of the current and previous frames. If pow_sum is lower than the constant POW_TONE_THR, tone-flag is set to zero.

The difference between the signal levels of the input frame and the background noise estimate is calculated as follows:

$$snr_sum = \sum_{n=1}^{12} MAX(1.0, \frac{level[n]}{bckr_est[n]})^2, \tag{6}$$

where:

level[n] signal level at band n

bckr_est[n] level of background noise estimate at band n

VAD decision is made by comparing the variable *snr_sum* to a threshold. The threshold (*vad_thr*) is adapted to get desired sensitivity depending on estimated speech and background noise levels.

Average background noise level is calculated by adding noise estimates at each band except the lowest band:

$$noise_level = \sum_{n=2}^{12} bckr_est[n] \quad (7)$$

If SNR is lower than the threshold (*MIN_SPEECH_SNR*), speech level is increased as follows:

```
If (speech_level/noise_level < MIN_SPEECH_SNR)
    Speech_level = MIN_SPEECH_SNR * noise_level
```

Logarithmic value for noise estimate is calculated as follows:

$$i \log_2_noise_level = \log_2(noise_level) \quad (8)$$

Before logarithmic value from the speech estimate is calculated, *MIN_SPEECH_SNR*noise_level* is subtracted from the speech level to correct its value in low SNR situations.

$$i \log_2_speech_level = \log_2(speech_level - MIN_SPEECH_SNR * noise_level) \quad (9)$$

Threshold for VAD decision is calculated as follows:

$$Vad_thr = NO_SLOPE * (i \log_2_noise_level - NO_P1) + THR_HIGH + MIN(SP_CH_MAX, MAX(SP_CH_MIN, SP_CH_MIN + SP_SLOPE * (i \log_2_speech_level - SP_P1))), \quad (10)$$

where *NO_SLOPE*, *SP_SLOPE*, *NO_P1*, *SP_P1*, *THR_HIGH*, *SP_CH_MAX* and *SP_CH_MIN* are constants.

The variable *vadreg* indicates intermediate VAD decision and it is calculated as follows:

```
if (snr_sum > vad_thr)
    vadreg = 1
else
    vadreg = 0
```

3.3.3.1 Hangover addition

Before the final VAD flag is given, a hangover is added. The hangover addition helps to detect low power endings of speech bursts, which are subjectively important but difficult to detect.

VAD flag is set to '1' if less than *hang_len* frames with '0' decision have been elapsed since *burst_len* consecutive '1' decisions have been detected. The variables *hang_len* and *burst_len* are computed using *vad_thr* as follows:

$$hang_len = MAX(HANG_LOW, (HANG_SLOPE * (vad_thr - HANG_P1) + HANG_HIGH)) \quad (11)$$

$$burst_len = BURST_SLOPE * (vad_thr - BURST_P1) + BURST_HIGH \quad (12)$$

The power of the input frame is compared to a threshold (*VAD_POW_LOW*). If the power is lower, the VAD flag is set to '0' and no hangover is added. The *VAD_flag* is calculated as follows:

```
Vad_flag = 0;
if (pow_sum < VAD_POW_LOW)
    burst_count = 0
    hang_count = 0
else
    if (vadreg = 1)
        burst_count = burst_count + 1
        if (burst_count >= burst_len)
            hang_count = hang_len
            VAD_flag = 1
```

```

else
  burst_count = 0
  if (hang_count > 0)
    hang_count = hang_count - 1
    VAD_flag=1

```

3.3.3.2 Background noise estimation

Background noise estimate ($bckr_est[n]$) is updated using amplitude levels of the previous frame. Thus, the update is delayed by one frame to avoid undetected start of speech bursts to corrupt the noise estimate. The update speed for the current frame is selected using intermediate VAD decisions ($vadreg$) and stationarity counter ($stat_count$) as follows:

```

if (vadreg for the last 4 frames has been zero)
  alpha_up = ALPHA_UP1
  alpha_down = ALPHA_DOWN1
else if (stat_count = 0)
  alpha_up = ALPHA_UP2
  alpha_down = ALPHA_DOWN2
else
  alpha_up = 0
  alpha_down = ALPHA3

```

The variable $stat_count$ indicates stationary and its purpose is explained later in this subclause. The variables $alpha_up$ and $alpha_down$ define the update speed for upwards and downwards, respectively. The update speed for each band "n" is selected as follows:

```

if ( $bckr\_est_m[n] < level_{m-1}[n]$ )
  alpha[n] = alpha_up
else
  alpha[n] = alpha_down

```

Finally, noise estimate is updated as follows:

$$bckr_est_{m+1}[n] = (1.0 - alpha[n]) * bckr_est_m[n] + alpha[n] * level_{m-1}[n], \quad (13)$$

where:

- n index of the frequency band
- m index of the frame

Level of the background estimate ($bckr_est[n]$) is limited between constants NOISE_MIN and NOISE_MAX.

If level of background noise increases suddenly, $vadreg$ will be set to "1" and background noise is not normally updated upwards. To recover from this situation, update of the background noise estimate is enabled if the intermediate VAD decision ($vadreg$) is '1' for long enough time and spectrum is stationary. Stationary ($stat_rat$) is estimated using following equation:

$$stat_rat = \sum_{n=1}^{12} \frac{MAX(STAT_THR_LEVEL, MAX(ave_level_m[n], level_m[n]))}{MAX(STAT_THR_LEVEL, MIN(ave_level_m[n], level_m[n]))}, \quad (14)$$

where:

- STAT_THR_LEVEL a constant
- n index of the frequency band
- m index of the frame
- ave_level average level of the input signal

If the stationary estimate (*stat_rat*) is higher than a threshold, the stationary counter (*stat_count*) is set to the initial value defined by constant *STAT_COUNT*. If the signal is not stationary but speech has been detected (VAD decision is '1'), *stat_count* is decreased by one in each frame until it is zero.

```

if (5 last tone flags have been one)
    stat_count = STAT_COUNT
else
    if (8 last internal VAD decisions have been zero) OR (stat_rat > STAT_THR)
        stat_count = STAT_COUNT
    else
        if (vadreg) AND (stat_count ≠ 0)
            stat_count = stat_count - 1

```

The average signal levels (*ave_level[n]*) are calculated as follows:

$$ave_level_{m+1}[n] = (1.0 - alpha) * ave_level_m[n] + alpha * level_m[n] \quad (15)$$

The update speed (*alpha*) for the previous equation is selected as follows:

```

if (stat_count = STAT_COUNT)
    alpha = 1.0
else if (vadreg = 1)
    alpha = ALPHA5
else
    alpha = ALPHA4

```

3.3.3.3 Speech level estimation

First, full-band input level is calculated by summing input levels in each band except the lowest band as follows:

$$in_level = \sum_{n=2}^{12} level[n] \quad (16)$$

A frame is assumed to contain speech if its level is high enough (*MIN_SPEECH_LEVEL1*), and the intermediate VAD flag (*vadreg*) is set or the input level is higher than the current speech level estimate. Maximum level (*sp_max*) from *SP_EST_COUNT* frames is searched. If the *SP_ACTIVITY_COUNT* number of speech frames is located in within *SP_EST_COUNT* number of frames, speech level estimate is updated by the maximum signal level (*sp_max*). The pseudocode for the speech level estimation is as follows:

```

If (SP_ACTIVITY_COUNT > SP_EST_COUNT - sp_est_cnt + sp_max_cnt)
    sp_est_cnt = 0
    sp_max_cnt = 0
    sp_max = 0
sp_est_cnt = sp_est_cnt + 1
if (in_level > MIN_SPEECH_LEVEL1) AND ((vadreg = 1) OR (in_level > speech_level))
    sp_max_cnt = sp_max_cnt + 1
    sp_max = MAX(sp_max, in_level)
    if (sp_max_cnt > SP_ACTIVITY_COUNT)
        if (sp_max > MIN_SPEECH_LEVEL2)
            if (sp_max > speech_level)
                speech_level = speech_level + SP_ALPHA_UP * (sp_max - speech_level)
            else
                speech_level = speech_level + SP_ALPHA_DOWN * (sp_max - speech_level)
        sp_max_cnt = 0
        sp_max = 0
    sp_est_cnt = 0

```

4 Computational details

A low level description has been prepared in form of ANSI C-code [1].

Annex A (informative) :

Change history

Change history							
Date	TSG #	TSG Doc.	CR	Rev	Subject/Comment	Old	New
03-2001	11	SP-010089			Version 2.0.0 presented for approval		5.0.0
12-2004	26				Version for Release 6	5.0.0	6.0.0
06-2007	36				Version for Release 7	6.0.0	7.0.0
12-2008	42				Version for Release 8	7.0.0	8.0.0
12-2009	46				Version for Release 9	8.0.0	9.0.0
03-2011	51				Version for Release 10	9.0.0	10.0.0
02-2012					Edithelp improvements	10.0.0	10.0.1

History

Document history		
V10.0.0	April 2011	Publication (withdrawn)
V10.0.1	March 2012	Publication