Technical Specification Group Services and System Aspects
Meeting #6, Nice, France, December 15-17, 1999

TSGS#6(99)570
Agenda Item: 5.4.3

Source **TSG-S4**
Title **CRs on AMR**

S4 Tdoc.	Spec.	Ver.	CR	Rev.	Rel.	Subject
S4-99456	06.73	7.2.0	A020		R98	Correction to reset function in AMR decoder
S4-99523	06.75	7.0.0	A001		R98	Update of AMR Transmission Delay Figures
S4-99454R	06.93	7.2.0	A006	1	R98	Editorial clarifications concerning RATSCCH and RX/TX DTX handler synchronization at handover.
S4-99455R	06.93	7.2.0	A007	1	R98	Onset frame signaling by the TX RSS.
S4-99380	26.090	3.0.1	001		R99	Bit allocation of the adaptive multi-rate codec
S4-99392	26.090	3.0.1	001		R99	Use of random excitation when RX_NODATA and not in DTX.
S4-99538	26.093	3.0.1	001	2	R99	Alignment to GSM 06.93

Joint SMG11/TSG-SA4 Meeting #13 / #8
Kyoto, Japan, 6-10 December 1999

Document S4-99456
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

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06.73	**CR**	**A020**	Current Version:	7.2.0 (R98)

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	TSG-SA#6	for approval	X	strategic		(for SMG use only)
list expected approval meeting # here ↑		for information		non-strategic	X	

Form: CR cover sheet, version 2 for 3GPP and SMG The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	X	UTRAN / Radio		Core Network	X

Source:	TSG-SA WG4 Codec	**Date:**	16.12.1999
Subject:	Correction to reset function in AMR decoder		
Work item:	AMR		

Category:				**Release:**		
	F	Correction	X		Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature			Release 97	
shall be marked	C	Functional modification of feature			Release 98	X
with an X)	D	Editorial modification			Release 99	
					Release 00	

Reason for change:	The variables corr_hp and lsf_hist_mean_index are not used. In Decoder_amr_reset, a call to cb_gain_average_reset is missing. Table 7 and 8 (list of static variables) does not fully conform to C code.
Clauses affected:	C code files vad1.h, dtx_dec.h, dec_amr.c, and specification

Other specs affected:			
	Other 3G core specifications		→ List of CRs:
	Other GSM core specifications		→ List of CRs:
	MS test specifications		→ List of CRs:
	BSS test specifications		→ List of CRs:
	O&M specifications		→ List of CRs:

Other comments:	

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<---------- double-click here for help and instructions on how to create a CR.

1. How the code is changed

1.1 File vad1.h

Line 68 is removed (corr_hp is not used).

1.1.1 Before the change (line 65-69)

```
Word16 complex_hang_count;   /* complex hangover counter, used by VAD    */
Word16 complex_hang_timer;   /* hangover initiator, used by CAD          */

Word16 corr_hp;              /* filtered value                           */
Word16 best_corr_hp;         /* FIP filtered value Q15                   */
```

1.1.2 After the change (line 65-68)

```
Word16 complex_hang_count;   /* complex hangover counter, used by VAD    */
Word16 complex_hang_timer;   /* hangover initiator, used by CAD          */

Word16 best_corr_hp;         /* FIP filtered value Q15                   */
```

1.2 File dtx_dec.h

Line 55 is removed (lsf_hist_mean_index is not used).

1.2.1 Before the change (line 54-56)

```
Word16 lsf_hist_mean[M*DTX_HIST_SIZE];
Word16 lsf_hist_mean_index;
Word16 log_pg_mean;
```

1.2.2 After the change (line 54-55)

```
Word16 lsf_hist_mean[M*DTX_HIST_SIZE];
Word16 log_pg_mean;
```

1.3 File dec_amr.c

In Decoder_amr_reset a call to Cb_gain_average_reset is added.

1.3.1 Before the change (line 192-196)

```
for (i = 0; i < 9; i++)
   state->ltpGainHistory[i] = 0;

if (mode != MRDTX)
   lsp_avg_reset(state->lsp_avg_st);
```

1.3.2 After the change (line 192-198)

```
for (i = 0; i < 9; i++)
   state->ltpGainHistory[i] = 0;

Cb_gain_average_reset(state->Cb_gain_averageState);
```

```
if (mode != MRDTX)
   lsp_avg_reset(state->lsp_avg_st);
```

2. How the specification is changed

4.5.3 Static variables used in the C-code

In this section two tables that specify the static variables for the speech encoder and decoder respectively are shown. All static variables are declared within a C **struct.**

Table 7: Speech encoder static variables

Struct name	Variable	Type[Length]	Description
Speech_Encode_ FrameState	cod_amr_state	cod_amrState	see below in this table
	pre_state	Pre_ProcessState	see below in this table
	dtx	Flag	Is set if DTX functionality is used
	complexityCounter	int	Used for wMOPS counting
Pre_ProcessState	y2_hi	Word16	filter state, upper word
	y2_lo	Word16	filter state, lower word
	y1_hi	Word16	filter state, upper word
	y1_lo	Word16	filter state, lower word
	x0	Word16	filter state
	x1	Word16	filter state
cod_amrState	old_speech	Word16[320]	speech buffer
	speech	Word16*	pointer to current frame in old_speech
	p_window	Word16*	pointer to LPC analysis window in old_speech
	p_window_12k2	Word16*	pointer to LPC analysis window with no lookahead in old_speech (MR122)
	new_speech	Word16*	pointer to the last 160 speech samples in old_speech
	old_wsp	Word16[303]	buffer holding spectral weighted speech
	wsp	Word16*	pointer to the current frame in old_wsp
	old_lags	Word16[5]	open loop LTP states
	ol_gain_flg	Word16[2]	enables open loop pitch lag weighting (MR102)
	old_exc	Word16[314]	excitation vector
	exc	Word16*	current excitation
	ai_zero	Word16[51]	history of weighted synth. filter followed by zero vector
	zero	Word16*	zero vector
	h1	Word16*	impulse response of weighted synthesis filter
	hvec	Word16[80]	zero vector followed by impulse response
	lpcSt	lpcState	see below in this table
	lspSt	lspState	see below in this table
	clLtpSt	clLtpState	see below in this table
	gainQuantSt	gainQuantState	see below in this table
	pitchOLWghtSt	pitchOLWghtState	see below in this table
	tonStabSt	tonStabState	see below in this table
	vadSt~~1~~	vadState1	see below in this table
	~~V~~vadSt~~2~~	~~V~~vadState2	see below in this table
	dtx	Flag	is set if DTX functionality is used
	dtx_encSt	dtx_encState	see below in this table
	mem_syn	Word16[10]	synthesis filter memory
	mem_w0	Word16[10]	weighting filter memory (applied to error signal)
	mem_w	Word16[10]	weighting filter memory (applied to input signal)
	mem_err	Word16[50]	filter memory for production of error vector
	error	Word16*	error signal (input minus synthesized speech)
	sharp	Word16	pitch sharpening gain
~~V~~vadState1	bckr_est	Word16[9]	background noise estimate
	ave_level	Word16[9]	averaged input components for stationary estimation
	old_level	Word16[9]	input levels of the previous frame
	sub_level	Word16[9]	input levels calculated at the end of a frame (lookahead)
	a_data5	Word16[6]	memory for the filter bank
	a_data3	Word16[5]	memory for the filter bank
	burst_count	Word16	counts length of a speech burst
	hang_count	Word16	hangover counter
	~~false_count~~	~~Word16~~	~~False speech detection counter~~
	stat_count	Word16	stationary counter
	vadreg	Word16	15 flags for intermediate VAD decisions
	pitch	Word16	15 flags for pitch detection
	tone	Word16	15 flags for tone detection
	complex_high	Word16	flags for complex detection
	complex_low	Word16	flags for complex detection
	oldlag_count	Word16	variables for pitch detection
	oldlag	Word16	variables for pitch detection
	complex_hang_count	Word16	complex hangover counter, used by VAD
	complex_hang_timer	Word16	hangover initiator, used by CAD
	~~corr_hp~~	~~Word16~~	~~filtered value~~
	best_corr_hp	Word16	filtered value
	speech_vad_decision	Word16	final decision
	complex_warning	Word16	complex background warning
	sp_burst_count	Word16	counts length of a speech burst incl HO addition
	corr_hp_fast	Word16	filtered value

(continued)

Table 7 (concluded): Speech encoder static variables

Struct name	Variable	Type[Length]	Description
~~V~~vadState2	pre_emp_mem	Word16	input pre-emphasis memory
	update_cnt	Word16	noise update counter
	hyster_cnt	Word16	hysteresis counter
	last_update_cnt	Word16	noise update counter value for last frame
	ch_enrg_long_db	Word16[16]	long term channel energy in dB
	Lframe_cnt	Word32	10 ms frame counter
	Lch_enrg	Word32[16]	channel energy estimate
	Lch_noise	Word32[16]	channel noise estimate
	last_normb_shift	Word16	block shift factor for last frame, used for pre_emp_mem
	tsnr	Word16	total estimated peak SNR in dB
	hangover	Word16	VAD hangover
	burstcount	Word16	number of consecutive voice active frames
	fupdate_flag	Word16	A flag to control a forced update of the noise estimate
	negSNRvar	Word16	SNR variability
	negSNRbias	Word16	sensitivity bias
	shift_state	Word16	indicates scaling state of channel energy estimate
	L_R0	Word32	LTP energy
	L_Rmax	Word32	LTP max correlation
	LTP_flag	Flag	set when open loop pitch prediction gain > threshold
dtx_encState	lsp_hist	Word16[80]	LSP history (8 frames)
	log_en_hist	Word16[8]	logarithmic frame energy history (8 frames)
	hist_ptr	Word16	pointer to the cyclic history vectors
	log_en_index	Word16	Index for logarithmic energy
	~~hist_ptr_tmp~~	~~Word16~~	~~logarithmic frame energy index~~
	init_lsf_vq_index	Word16	initial index for lsf predictor
	lsp_index	Word16[3]	lsp indecies to the three code books
	dtxHangoverCount	Word16	is decreased in DTX hangover period
	decAnaElapsedCount	Word16	counter for elapsed speech frames in DTX
lpcState	LevinsonSt	LevinsonState	see below
LevinsonState	old_A	Word16[11]	last frames direct form coefficients
lspState	lsp_old	Word16[10]	old LSP vector
	lsp_old_q	Word16[10]	old quantized LSP vector
	qSt	Q_plsfState	see below in this table
Q_plsfState	past_rq	Word16[10]	past quantized LSF prediction error
clLtpState	pitchSt	Pitch_frState	see below in this table
tonStabState	count	Word16	count consecutive (potential) resonance frames
	gp	Word16[7]	pitch gain history
Pitch_frState	T0_prev_subframe	Word16	integer. pitch lag of previous subframe
gainQuantState	sf0_exp_gcode0	Word16	subframe 0/2 codebook gain exponent
	sf0_frac_gcode0	Word16	subframe 0/2 codebook gain fraction
	sf0_exp_target_en	Word16	subframe 0/2 target energy exponent
	sf0_frac_target_en	Word16	subframe 0/2 target energy fraction
	sf0_exp_coeff	Word16[5]	subframe 0/2 energy coefficient exponents
	sf0_frac_coeff	Word16[5]	subframe 0/2 energy coefficient fractions
	gain_idx_ptr	Word16*	pointer to gain index value in parameter frame
	gc_predSt	gc_predState	see below in this table
	gc_predUncSt	gc_predState	see below in this table
	adaptSt	GainAdaptState	see below in this table
gc_predState	past_qua_en	Word16[4]	MA predictor memory (20*log10(pred. error))
	past_qua_en_MR122	Word16[4]	MA predictor memory, 12.2 style (log2(pred. error))
GainAdaptState	onset	Word16	onset counter
	prev_alpha	Word16	previous adaptor output
	prev_gc	Word16	previous codebook gain
	ltpg_mem	Word16[5]	pitch gain history
pitchOLWghtState	old_T0_med	Word16	weighted open loop pitch lag
	ada_w	Word16	weigthing level depeding on open loop pitch gain
	wght_flg	Word16	switches lag weighting on and off

Table 8: Speech decoder static variables

Struct name	Variable	Type[Length]	Description
Speech_Decode_FrameState	~~D~~decoder_amrState	Decoder_amrState	see below in this table
	post_state	Post_FilterState	see below in this table
	~~P~~postHP_state	Post_ProcessState	see below in this table
	ComplexityCounter	int	Used for wMOPS counting
Decoder_amrState	old_exc	Word16[194]	excitation vector
	exc	Word16*	current excitation
	lsp_old	Word16[10]	LSP vector of previous frame
	mem_syn	Word16[10]	synthesis filter memory
	sharp	Word16	pitch sharpening gain
	old_T0	Word16	pitch sharpening lag
	prev_bf	Word16	previous value of “bad frame” flag
	prev_pdf	Word16	previous value of “pot. dangerous frame” flag
	state	Word16	ECU state (0..6)
	excEnergyHist	Word16[9]	excitation energy history
	T0_lagBuff	Word16	received pitch lag for ECU
	inBackgroundNoise	Word16	background noise flag
	voicedHangover	Word16	hangover flag
	ltpGainHistory	Word16[9]	pitch gain history
	background_state	Bgn_scdState	see below in this table
	Cb_gain_averState	Cb_gain_averageState	see below in this table
	lsp_avg~~State~~_st	lsp_avgState	see below in this table
	lsfState	D_plsfState	see below in this table
	ec_gain_p_st	ec_gain_pitchState	see below in this table
	ec_gain_c_st	ec_gain_codeState	see below in this table
	pred_state	gc_predState	see table 7
	nodataSeed	Word16	seed for CN generator
	ph_disp_st	ph_dispState	see below in this table
	dtxDecoderState	dtx_decState	see below in this table
dtx_decState	since_last_sid	Word16	number of frames since last SID frame
	true_sid_period_inv	Word16	inverse of true SID update rate
	log_en	Word16	logarithmic frame energy
	old_log_en	Word16	previous value of log_en
	L_pn_seed_rx	Word32	random number generator seed
	lsp	Word16[10]	LSP vector
	lsp_old	Word16[10]	previous LSP vector
	lsf_hist	Word16[80]	LSF vector history (8 frames)
	lsf_hist_ptr	Word16	index to beginning of LSF history
	lsf_hist_mean	Word16[80]	mean-removed LSF history (8 frames)
	~~lsf_hist_mean_index~~	~~Word16~~	~~index to beg. of mean-removed LSF history~~
	log_pg_mean	Word16	mean-removed logarithmic prediction gain
	log_en_hist	Word16[8]	logarithmic frame energy history
	log_en_hist_ptr	Word16	index to beginning of log, frame energy history
	log_en_adjust	Word16	mode-dependent frame energy adjustment
	dtxHangoverCount	Word16	counts down in hangover period
	decAnaElapsedCount	Word16	counts elapsed speech frames after DTX
	sid_frame	Word16	flags SID frames
	valid_data	Word16	flags SID frames containing valid data
	dtxHangoverAdded	Word16	flags hangover period at end of speech
	dtxGlobalState	enum DTXStateType	DTX state flags
	data_updated	Word16	flags CNI updates
Bgn_scdState	frameEnergyHist	Word16[60]	history of synthesis frame energy
	bgHangover	Word16	number of frames since last speech frame
Cb_gain_averageState	cbGainHistory	Word16[7]	codebook gain history
	hangVar	Word16	counts length of talkspurt in subframes
	hangCount	Word16	number of subframes since last talkspurt
lsp_avgState	lsp_meanSave	Word16[10]	averaged LSP vector
D_plsfState	past_r_q	Word16[10]	past quantized LSF prediction vector
	past_lsf_q	Word16[10]	past dequantized LSF vector
ec_gain_pitchState	pbuf	Word16[5]	pitch gain history
	past_gain_pit	Word16	previous pitch gain (limited to 1.0)
	prev_gp	Word16	previous good pitch gain
ec_gain_codeState	gbuf	Word16[5]	codebook gain history
	past_gain_code	Word16	previous codebook gain
	prev_gc	Word16	previous good codebook gain

(continued)

Table 8 (concluded): Speech decoder static variables

Struct name	Variable	Type[Length]	Description
ph_dispState	gainMem prevState prevCbGain lockFull onset	Word16[5] Word16 Word16 Word16 Word16	pitch gain history previously used impulse response previous codebook gain force maximum phase dispersion onset counter
Post_FilterState	res2 mem_syn_pst synth_buf agc_state preemph_state	Word16[40] Word16[10] Word16[170] agcState preemphasisState	LP residual synthesis filter memory synthesis filter work area see below in this table see below in this table
agcState	past_gain	Word16	past agc gain
preemphasisState	mem_pre	Word16	filter state
Post_ProcessState	y2_hi y2_lo y1_hi y1_lo x0 x1	Word16 Word16 Word16 Word16 Word16 Word16	filter state, upper word filter state, lower word filter state, upper word filter state, lower word filter state filter state

SMG11#13/TSG-S4$8 Meeting
Kyoto, Japan,December 6-10, 1999

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e.g. for 3GPP use the format TP-99xxx
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06.75 CR A001 Current Version: 7.0.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	TSG-SA#6	for approval	X	strategic		*(for SMG use only)*
list expected approval meeting # here ↑		for information		non-strategic	X	

Form: CR cover sheet, version 2 for 3GPP and SMG The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: (U)SIM [ ] ME [ ] UTRAN / Radio [X] Core Network [ ]
(at least one should be marked with an X)

Source: TSG-S4/SMG11 **Date:** 16/10/99

Subject: Update of AMR Transmission Delay Figures

Work item: AMR

Category: *(only one category shall be marked with an X)*

F	Correction	X	**Release:**	Phase 2	
A	Corresponds to a correction in an earlier release			Release 96	
B	Addition of feature			Release 97	
C	Functional modification of feature			Release 98	X
D	Editorial modification			Release 99	
				Release 00	

Reason for change: The proposed modifications to the AMR transmission delay figures are based on the approved AMR TRAU frame formats, as defined in 08.60 and 08.61.

Clauses affected: 14

Other specs affected:

Other 3G core specifications		→ List of CRs:	
Other GSM core specifications		→ List of CRs:	
MS test specifications		→ List of CRs:	
BSS test specifications		→ List of CRs:	
O&M specifications		→ List of CRs:	

Other comments:

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<--------- double-click here for help and instructions on how to create a CR.

14 Transmission Delay

~~*Editor's Note 1: Section based on the content of Tdoc SMG11 158/99. This document was produced before a final agreement was reached on the format of the Abis and Ater TRAU frames. The final format could have a [small] impact on the delay figures presented below.*~~

The transmission delay of a communication using AMR has been evaluated using the same method as for the previous GSM speech codecs [2, 3 & 4]. The reference system delay distribution for the downlink and uplink directions are provided in figures 14.1 and 14.2 respectively. The speech transcoders are assumed to be remote located from the BTS (16 kbit/s or 8 kbit/s sub-multiplexing on the Abis & Ater Interfaces).

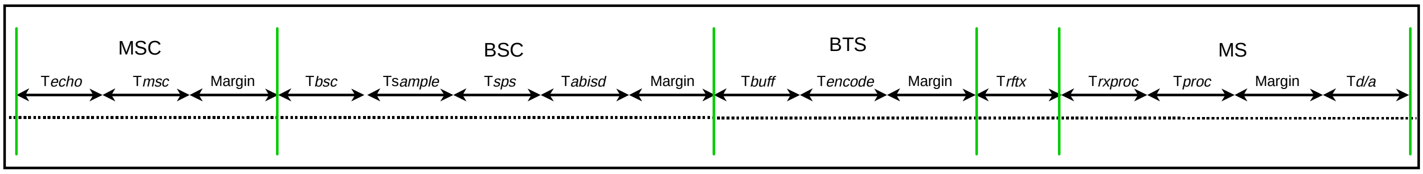

Figure 14.1: Reference Downlink delay distribution

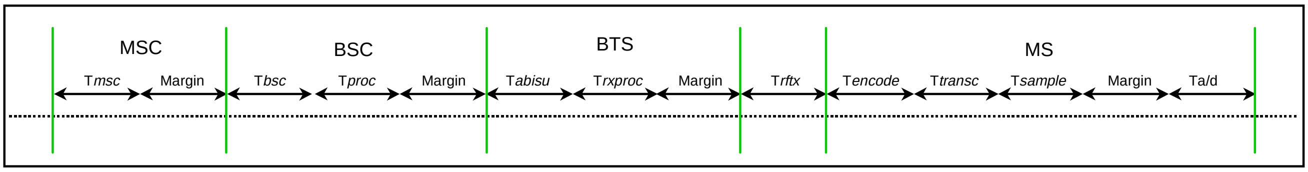

Figure 14.2: Reference Uplink delay distribution

The definition of the different delay parameters is given in the following table. The table also provides the value used for the parameter when not dependent of the type of speech codec or sub-multiplexing scheme over the Abis & Ater interfaces.

T*abisd* Time required to transmit the minimum number of speech data bits over the downlink Abis interface to start encoding a radio speech frame. Depends on the speech codec mode, the TRAU frame format and the Abis/Ater sub-multiplexing scheme. Note that most TRAU frame synchronization bits can ideally be transmitted by anticipation and are usually not included in this parameter.

T*abisu* Time required to transmit the minimum number of speech data bits over the uplink Abis interface to start decoding a speech frame. Depends on the speech codec mode, the TRAU frame format and the Abis/Ater sub-multiplexing scheme. Note that the TRAU frame synchronization bits can ideally be transmitted by anticipation and are usually not included in this parameter.

T*a/d* Delay in the analogue to digital converter in the uplink (implementation dependent). Set to 1ms [4].

T*bsc* Switching delay in the BSC (implementation dependent). Set to 0.5ms [2 & 4].

T*buff* Buffering time required for the time alignment procedure for the in-band control of the remote transcoder. Set to 1.25 ms [2 & 4].

T*d/a* Delay in the digital to analogue converter in the downlink (implementation dependent). Set to 1ms [2 & 4].

T*echo* Delay induced by the echo canceller (implementation dependent). Set to 1ms [2 & 4].

T*encode*: Processing delay required to perform the channel encoding (implementation dependent). Depends on the channel coding complexity of each codec mode.

T*msc* Switching delay in the MSC (implementation dependent). Set to 0.5ms [2 & 4].

T*proc* Processing delay required to perform the speech decoding (implementation dependent). Depends on the speech decoding complexity of each codec mode.

T*rftx* Time required for the transmission of a speech frame over the air interface. Derived from the radio framing structure and the interleaving scheme. Worst case is 37.5 ms in Full Rate mode and 32.5 ms in Half Rate mode [2 & 4].

T*rxproc* Processing delay required to perform the channel equalization, the channel decoding and SID-frame detection (implementation dependent). The channel decoding depends on the codec mode. The channel equalization part was set to 6.84 ms in Full Rate mode and 3.5 ms in Half Rate mode [4].

T*sample* Duration of the segment of PCM speech samples operated on by the speech transcoder: 25 ms in all cases corresponding to 20 ms for the processed speech frame and 5 ms of look ahead.

T*sps* Worst case processing delay required by the downlink speech encoder before an encoded bit can be sent over the Ater/Abis interface taking into account the speed on the Ater/Abis interface (implementation dependent). Depends on the speech coding complexity of each codec mode and on the sub-multiplexing rate on the Ater/Abis interface. Because of the priority given to the decoding, T*proc* is also added to the overall downlink transmission delay.

T*transc* MS speech encoder processing delay, from input of the last PCM sample to output of the final encoded bit (implementation dependent). For the evaluation of the transmission delay, it was assumed that the speech decoding has a higher priority than the speech encoding, i.e. this delay is artificially increased by the speech decoding delay.

Margin Implementation dependent margins in the different system components. Set as follows:
MSC Margin: 0.5 ms [2 & 4]
BSC Margin: 0.5 ms [2 & 4]
BTS Margin: 0.45 ms downlink, 0.3 ms uplink [2 & 4]
MS Margin: 2 ms in Full Rate, 1.9 ms in Half Rate [2 & 4].

The processing delays were estimated using complexity figures for each codec mode. In addition, to take into account the dependence on the DSP implementation, the computation was based on the same methodology used for the previous GSM speech codecs [4].

The DSPs running the speech and channel codec are modeled with the 3 following parameters:

E represents the DSP Efficiency. This corresponds to the ratio tMOPS/wMOPS of the codec implementation on the DSP.

S represents for the speed of the DSP: Maximum Number of Operations that the DSP can run in 1 second. This number is expressed in MOPS.

P represents the percentage of DSP processing power assigned to the codec.

The processing delay of a task of complexity X (in wMOPS) can then be computed using the equation:

$$D = \frac{20X}{ESP} \text{ ms}$$

For compatibility reasons, the same ESP parameter used for the EFR processing delays computation [4] was used: ESP=25[1].

The following tables provide the overall transmission delay parameters for each codec mode. The design objective for the Algorithmic Round Trip Transmission Delay (ARTD = 2T*sample* + 2T*rftx* + T*abisu* + T*abisd*) was set to the EFR ARTD increased by 10 ms in Full Rate mode, and the GSM HR ARTD increased by 10 ms in Half Rate mode.

Tables 14.1 and 14.2 define the parameters impacting the computation of the transmission delays over the Abis/Ater interfaces (T*abisu* & T*abisd*) for the 16 kbit/s and 8kbit/s sub-multiplexing schemes respectively. The definition of different parameters is provided below. They are derived from the AMR TRAU frame format provided in [5 & 6].

Min # of bits: Minimum number of speech bits required to start the next operation (speech decoding in uplink or channel encoding in downlink).

Sync. bits: Additional synchronization bits in the TRAU frame (synchronization header not included) before reaching the last required bit.

Min # Data: Rank of the last required bit in the TRAU frame.

 # Anticip.: Number of bits that can be sent by anticipation.

---

1 This ESP value was derived in 1996, during the EFR standardization. It is based on a 40 MHz DSP, with an efficiency of 1 and a 60% CPU availability. All processing delays would be improved assuming DSP performances corresponding to the state of the art of DSP technology.

 # Requir.: Resulting number of bits that must be received (Min #Data - # Anticip.).

	Mode	Min # of bits	Sync. bits	Min # Data	# anticip.	# Requir.	Tabisu	Mode	Min # of bits	Sync. bits	Min # Data	# anticip.	# Requir.	Tabisu
Full	**12.2**		6	143	38	105	6.625	**12.2**	94	6	139	33	106	6.625
Rate	**10.2**		6	144	38	106	6.625	**10.2**	75	5	119	33	86	5.375
16k	**7.95**		6	144	38	106	6.625	**7.95**	64	4	107	33	74	4.625
Upl	**7.4**		6	144	38	106	6.625	**7.4**	61	4	104	33	71	4.5
	6.7		6	144	38	106	6.625	**6.7**	58	4	101	33	70	4.375
	5.9		6	144	38	106	6.625	**5.9**	54	3	96	33	65	4.125
	5.15		6	144	38	106	6.625	**5.15**	49	3	91	33	60	3.75
	4.75		6	144	38	106	6.625	**4.75**	51	3	93	33	62	3.875

	Mode	Min # of bits	Min # Data	# anticip.	# Requir.	Tabisd	Mode	Min # of bits	Min # Data	# anticip.	# Requir.	Tabisd
Full	**12.2**		316	43	273	17.125	**12.2**	256	312	39	273	17.125
Rate	**10.2**		316	43	273	17.125	**10.2**	216	312	82	230	14.375
16k	**7.95**		259	43	216	13.5	**7.95**	171	312	130	182	11.375
Dwnl	**7.4**		250	43	207	13	**7.4**	160	312	141	171	10.75
	6.7		238	43	195	12.25	**6.7**	146	312	156	156	9.75
	5.9		230	43	187	11.75	**5.9**	130	312	173	139	8.75
	5.15		215	43	172	10.75	**5.15**	115	312	189	123	7.75
	4.75		204	43	161	10.125	**4.75**	107	312	198	114	7.125

Table 14.1: Tabisu (ms) & Tabisd (ms) computation tables for the 16 kbit/s sub-multiplexing scheme

	Mode	Min # of bits	Sync. bits	Min # Data	# anticip.	# Requir.	Tabisu		Mode	Min # of bits	Sync. bits	Min # Data	# anticip.	# Requir.	Tabisu
Half	**7.95**	-	-	-	-	-	-	Half	**7.95**	-	-	-	-	-	-
Rate	**7.4**			70	3	67	8.375	Rate	**7.4**	58	0	72	2	70	8.75
8k	**6.7**			76	9	67	8.375	8k	**6.7**	55	0	75	12	63	7.875
Upl	**5.9**			77	17	60	7.5	Upl	**5.9**	51	0	77	12	65	8.125
	5.15			77	22	55	6.875		**5.15**	46	0	68	12	56	7
	4.75			77	20	57	7.125		**4.75**	48	0	70	12	58	7.25

	Mode	Min # of bits	Min # Data	# anticip.	# Requir.	Tabisd		Mode	Min # of bits	Min # Data	# anticip.	# Requir.	Tabisd
Half	**7.95**	-	-	-	-	-	Half	**7.95**	-	-	-	-	-
Rate	**7.4**		160	3	157	19.625	Rate	**7.4**	148	174	12	162	20.25
8k	**6.7**		160	9	151	18.875	8k	**6.7**	134	160	22	138	17.25
Dwnl	**5.9**		158	17	141	17.625	Dwnl	**5.9**	118	160	22	138	17.25
	5.15		157	22	135	16.875		**5.15**	103	160	39	121	15.125
	4.75		147	20	127	15.875		**4.75**	95	160	47	113	14.125

Table 14.2: Tabisu (ms) & Tabisd (ms) computation tables for the 8 kbit/s sub-multiplexing scheme

Tables 14.3 and 14.4 provide the overall Uplink and Downlink transmission delay for the different Full Rate codec modes using a 16 kbit/s sub-multiplexing scheme.

Tables 14.5 and 14.6 provide the overall Uplink and Downlink transmission delay for the different Half Rate codec modes using a 16 kbit/s sub-multiplexing scheme.

Tables 14.7 and 14.8 provide the overall Uplink and Downlink transmission delay for the different Half Rate codec modes using an 8 kbit/s sub-multiplexing scheme.

UL FR 16k	Delay Parameter	**12.2**	**10.2**	**7.95**	**7.4**	**6.7**	**5.9**	**5.15**	**4.75**	**FR**	**EFR**
MSC	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	1.8160	1.8320	1.9920	1.7600	2.3600	2.024	2.0160	2.0160	1.5	1.27
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	6.625	6.625	6.625	6.6	6.625	6.625	6.63	6.625	4	6.4375
	Trxproc eq	6.84	6.84	6.84	6.84	6.84	6.84	6.84	6.84	8.8	6.84
	Trxproc ch. dec.	1.9360	1.7440	3.880	1.3280	0.2560	0.2400	0.232	0.2320	0	1.96
	Margin	3	3	3	3	3	3	3	3	3	3
	Trftx	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5
MS	Tencode	0.272	0.288	0.248	0.232	0.256	0.24	0.232	0.232	1.6	0.32
	Ttransc	12.976	12.680	13.256	12.104	13.50	11.0240	9.6560	11.240	8	12.17
	Tsample	25	25	25	25	25	25	25	25	20	20
	Tmargin	2	2	2	2	2	2	2	2	2	2
	Ta/d	1	1	1	1	1	1	1	1	1	1
	Total Uplink	101.0	100.5	103.3	99.4	100.3	97.5	96.1	97.7	89.4	94.5

UL FR 16k	Delay Parameter	**12.2**	**10.2**	**7.95**	**7.4**	**6.7**	**5.9**	**5.15**	**4.75**	**FR**	**EFR**
MSC	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	1.5632	1.5328	1.6072	1.5168	1.5888	1.564	1.5624	1.5664	1.5	1.27
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	6.625	5.375	4.625	4.5	4.375	4.125	3.75	3.875	4	6.4375
	Trxproc eq	6.84	6.84	6.84	6.84	6.84	6.84	6.84	6.84	8.8	6.84
	Trxproc ch. dec.	1.6848	1.5504	3.408	1.1704	1.1288	2.7224	0.936	2.2568	0	1.96
	Margin	3	3	3	3	3	3	3	3	3	3
	Trftx	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5
MS	Tencode	0.2376	0.2576	0.2168	0.2	0.2184	0.2016	0.2072	0.1952	1.6	0.32
	Ttransc	11.379	11.127	11.714	10.950	11.66	9.7912	8.6912	10.250	8	12.17
	Tsample	25	25	25	25	25	25	25	25	20	20
	Tmargin	2	2	2	2	2	2	2	2	2	2
	Ta/d	1	1	1	1	1	1	1	1	1	1
	Total Uplink	98.8	97.2	98.9	95.7	96.3	95.7	92.5	95.5	89.4	94.5

Table 14.3: Uplink Transmission Delay in Full Rate Mode (in ms & 16 kbit/s sub-multiplexing scheme)

DL FR 16k

	Delay Parameter	12.2	10.2	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR
MSC	Techo	1	1	1	1	1	1	1	1	1	1
	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	25	25	25	25	25	25	25	25	20	20
	Tsps	2.28	2.28	2.28	2.28	2.28	2.28	2.28	2.28	1.6	2.3
	Tproc (Tsps)	1.8160	1.8320	1.9920	1.7600	2.3600	2.024	2.0160	2.0160		
	Tabisd	17.125	17.125	13.5	13	12.25	11.75	10.75	10.125	17.4	17.375
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tbuff	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	0.272	0.288	0.248	0.232	0.256	0.24	0.232	0.232	1.6	1.6
	Margin	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5
MS	Trxproc eq	6.84	6.84	6.84	6.84	6.84	6.84	6.84	6.84	8.8	6.84
	Trxproc ch. dec.	1.936	1.744	3.88	1.328	0.256	0.24	0.232	0.232	0	1.96
	Tproc	1.816	1.832	1.992	1.76	2.36	2.024	2.016	2.016	1.5	1.27
	Margin	2	2	2	2	2	2	2	2	2	2
	Td/a	1	1	1	1	1	1	1	1	1	1
	Total Downlink	102.3	102.1	100.9	97.4	96.8	95.6	94.6	93.9	96.1	96.5

DL FR 16k

	Delay Parameter	12.2	10.2	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR
MSC	Techo	1	1	1	1	1	1	1	1	1	1
	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	25	25	25	25	25	25	25	25	20	20
	Tsps	2.28	2.28	2.28	2.28	2.28	2.28	2.28	2.28	1.6	2.3
	Tproc (Tsps)	1.5632	1.5328	1.6072	1.5168	1.5888	1.564	1.5624	1.5664		
	Tabisd	17.125	14.375	11.375	10.75	9.75	8.75	7.75	7.125	17.4	17.375
	~~Margin~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~
BTS	Tbuff	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	0.2376	0.2576	0.2168	0.2	0.2184	0.2016	0.2072	0.1952	1.6	1.6
	Margin	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5	37.5
MS	Trxproc eq	6.84	6.84	6.84	6.84	6.84	6.84	6.84	6.84	8.8	6.84
	Trxproc ch. dec.	1.6848	1.5504	3.408	1.1704	1.1288	2.7224	0.936	2.2568	0	1.96
	Tproc	1.5632	1.5328	1.6072	1.5168	1.5888	1.564	1.5624	1.5664	1.5	1.27
	Margin	2	2	2	2	2	2	2	2	2	2
	Td/a	1	1	1	1	1	1	1	1	1	1
	Total Downlink	101.5	98.6	97.5	94.5	93.6	94.1	91.3	92.0	96.1	96.5

Table 14.4: Downlink Transmission Delay in Full Rate Mode (in ms & 16 kbit/s sub-multiplexing scheme)

UL HR16k	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR	HR
MSC	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	1.9920	1.7600	2.3600	2.0240	2.0160	2.0160	1.5	1.27	1.71
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	6.625	6.6	6.625	6.625	6.63	6.625	4	6.4375	4.8125
	Trxproc eq	3.5	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	1.1040	1.0800	1.0000	0.9440	0.9120	2.1280	0	1.96	2.3
	Margin	3	3	3	3	3	3	3	3	3
	Trftx	32.9	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Tencode	0.152	0.152	0.144	0.136	0.144	0.136	1.6	0.32	0.16
	Ttransc	13.256	12.104	13.50	11.0240	9.6560	11.240	8	12.17	15.6
	Tsample	25	25	25	25	25	25	20	20	24.4
	Tmargin	1.9	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Ta/d	1	1	1	1	1	1	1	1	1
	Total Uplink	92.4	91.0	92.9	90.1	88.7	91.4	89.4	94.5	93.3

UL HR16k	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR	HR
MSC	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	1.6072	1.5168	1.5888	1.5640	1.5624	1.5664	1.5	1.27	1.71
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	4.625	4.5	4.375	4.125	3.75	3.875	4	6.4375	4.8125
	Trxproc eq	3.5	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	1.8928	0.9312	0.9104	0.8448	0.7984	0.7696	0	1.96	2.3
	Margin	3	3	3	3	3	3	3	3	3
	Trftx	32.9	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Tencode	0.1368	0.1328	0.1216	0.1144	0.128	0.1192	1.6	0.32	0.16
	Ttransc	11.714	10.950	11.66	9.7912	8.6912	10.250	8	12.17	15.6
	Tsample	25	25	25	25	25	25	20	20	24.4
	Tmargin	1.9	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Ta/d	1	1	1	1	1	1	1	1	1
	Total Uplink	89.3	87.3	88.0	85.7	84.2	85.9	89.4	94.5	93.3

Table 14.5: Uplink Transmission Delay in Half Rate Mode (in ms & 16 kbit/s sub-multiplexing scheme)

	DL HR 16k Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR	HR
MSC	Techo	1	1	1	1	1	1	1	1	1
	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	25	25	25	25	25	25	20	20	24.4
	Tsps	1.74	1.74	1.74	1.74	1.74	1.74	1.6	2.3	7.8
	Tproc (Tsps)	1.992	1.76	2.36	2.024	2.016	2.016	0	0	0
	Tabisd	13.5	13	12.25	11.75	10.75	10.125	17.4	17.375	8.375
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tbuff	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	0.152	0.152	0.144	0.136	0.144	0.136	1.6	1.6	0.16
	Margin	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	32.9	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Trxproc eq	3.5	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	1.104	1.08	1	0.944	0.912	2.128	0	1.96	2.3
	Tproc	1.9920	1.7600	2.3600	2.0240	2.0160	2.0160	1.5	1.27	1.71
	Margin	1.9	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Td/a	1	1	1	1	1	1	1	1	1
	Total Downlink	89.5	88.5	88.9	87.6	86.6	87.2	96.1	96.5	88.7

	DL HR 16k Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR	EFR	HR
MSC	Techo	1	1	1	1	1	1	1	1	1
	Tmsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	25	25	25	25	25	25	20	20	24.4
	Tsps	2.1508	2.1508	2.1508	2.1508	2.1508	2.1508	1.6	2.3	7.8
	Tproc (Tsps)	1.6072	1.5168	1.5888	1.564	1.5624	1.5664	0	0	0
	Tabisd	11.375	10.75	9.75	8.75	7.75	7.125	17.4	17.375	8.375
	~~Margin~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~
BTS	Tbuff	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	0.1368	0.1328	0.1216	0.1144	0.128	0.1192	1.6	1.6	0.16
	Margin	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	32.9	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Trxproc eq	3.5	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	1.8928	0.9312	0.9104	0.8448	0.7984	0.7696	0	1.96	2.3
	Tproc	1.6072	1.5168	1.5888	1.5640	1.5624	1.5664	1.5	1.27	1.71
	Margin	1.9	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Td/a	1	1	1	1	1	1	1	1	1
	Total Downlink	87.8	86.0	85.1	84.0	83.0	82.3	96.1	96.5	88.7

Table 14.6: Downlink Transmission Delay in Half Rate Mode (in ms & 16 kbit/s sub-multiplexing scheme)

UL HR8k

	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR/16k	EFR/16k	HR
MSC	Tmsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	-	1.7600	2.3600	2.0240	2.0160	2.0160	1.5	1.27	1.71
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	-	8.38	8.375	7.500	7	7.13	4	6.4375	9.75
	Trxproc eq	-	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	-	1.0800	1.0000	0.9440	0.9120	2.1280	0	1.96	2.3
	Margin	-	3	3	3	3	3	3	3	3
	Trftx	-	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Tencode	-	0.152	0.144	0.136	0.144	0.136	1.6	0.32	0.16
	Ttransc	-	12.104	13.50	11.0240	9.6560	11.240	8	12.17	15.6
	Tsample	-	25	25	25	25	25	20	20	24.4
	Tmargin	-	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Ta/d	-	1	1	1	1	1	1	1	1
	Total Uplink	N/A	92.8	94.7	90.9	88.9	91.9	89.4	94.5	98.2

UL HR8k

	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR/16k	EFR/16k	HR
MSC	Tmsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tproc	-	1.5168	1.5888	1.5640	1.5624	1.5664	1.5	1.27	1.71
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tabisu	-	8.75	7.875	8.125	7	7.25	4	6.4375	9.75
	Trxproc eq	-	3.5	3.5	3.5	3.5	3.5	8.8	6.84	3.5
	Trxproc ch. dec.	-	0.9312	0.9104	0.8448	0.7984	0.7696	0	1.96	2.3
	Margin	-	3	3	3	3	3	3	3	3
	Trftx	-	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Tencode	-	0.1328	0.1216	0.1144	0.128	0.1192	1.6	0.32	0.16
	Ttransc	-	10.950	11.66	9.7912	8.6912	10.250	8	12.17	15.6
	Tsample	-	25	25	25	25	25	20	20	24.4
	Tmargin	-	1.9	1.9	1.9	1.9	1.9	2	2	1.9
	Ta/d	-	1	1	1	1	1	1	1	1
	Total Uplink	N/A	91.6	91.5	89.7	87.5	89.3	89.4	94.5	98.2

Table 14.7: Uplink Transmission Delay in Half Rate Mode (in ms & 8 kbit/s sub-multiplexing scheme)

DL HR 8k	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR/16k	EFR/16k	HR
MSC	Techo	-	1	1	1	1	1	1	1	1
	Tmsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	-	25	25	25	25	25	20	20	24.4
	Tsps	-	1.61	1.61	1.61	1.61	1.61	1.6	2.3	4.3
	Tproc (Tsps)	-								
	Tabisd	-	19.625	18.875	17.625	16.875	15.875	17.4	17.375	17.5
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BTS	Tbuff	-	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	-	0.152	0.144	0.136	0.144	0.136	1.6	1.6	0.16
	Margin	-	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	-	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Trxproc eq	-	3.5	3.5	3.5	3.5	3.5	6.84	8.8	3.5
	Trxproc ch. dec.	-	1.08	1	0.944	0.912	2.128	1.96	0	2.3
	Tproc	-	1.7600	2.3600	2.0240	2.0160	2.0160	1.5	1.27	1.71
	Margin	-	1.9	1.99	1.9	1.9	1.9	2	2	1.9
	Td/a	-	1	1	1	1	1	1	1	1
	Total Downlink	N/A	93.2	93.1	91.3	90.6	90.8	96.1	96.5	94.4

DL HR 8k	Delay Parameter	7.95	7.4	6.7	5.9	5.15	4.75	FR/16k	EFR/16k	HR
MSC	Techo	-	1	1	1	1	1	1	1	1
	Tmsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Margin	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
BSC	Tbsc	-	0.5	0.5	0.5	0.5	0.5	0.5	0.5	0.5
	Tsample	-	25	25	25	25	25	20	20	24.4
	Tsps	-	10	10	10	10	10	1.6	2.3	4.3
	Tproc (Tsps)	-								
	Tabisd	-	20.25	17.25	17.25	15.125	14.125	17.4	17.375	17.5
	~~Margin~~	~~-~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~	~~0.5~~
BTS	Tbuff	-	1.25	1.25	1.25	1.25	1.25	1.25	1.25	1.25
	Tencode	-	0.1328	0.1216	0.1144	0.128	0.1192	1.6	1.6	0.16
	Margin	-	0.45	0.45	0.45	0.45	0.45	0.45	0.45	0.45
	Trftx	-	32.9	32.9	32.9	32.9	32.9	37.5	37.5	32.9
MS	Trxproc eq	-	3.5	3.5	3.5	3.5	3.5	6.84	8.8	3.5
	Trxproc ch. dec.	-	0.9312	0.9104	0.8448	0.7984	0.7696	1.96	0	2.3
	Tproc	-	1.5168	1.5888	1.5640	1.5624	1.5664	1.5	1.27	1.71
	Margin	-	1.9	1.99	1.9	1.9	1.9	2	2	1.9
	Td/a	-	1	1	1	1	1	1	1	1
	Total Downlink	N/A	101.8	99.0	98.8	96.6	95.6	96.1	96.5	94.4

Table 14.8: Downlink Transmission Delay in Half Rate Mode (in ms & 8 kbit/s sub-multiplexing scheme)

15 Frequency Response

NOTE: The frequency response is essentially given as a piece of additional information. It should not be used to qualify the codec performances in terms of perceived quality or DTMF transparency.

The frequency response of the AMR codec was evaluated by computing the logarithmic gain of the frequency response of each codec mode, according to the following equation:

$$\mathrm{Gain}_{dB} = 10\log_{10}\left[\sum_{k=1}^{M}\mathrm{out}(k)^2 / \sum_{k=1}^{M}\mathrm{inp}(k)^2\right]$$

where inp(k) and out(k) are the input (original) and output (processed) signals and M is the total number of processed samples.

The frequency response was computed for all 8 codec modes (12.2, 10.2, 7.95, 7.4, 6.7, 5.9, 5.15 and 4.75 kbit/s), in error-free condition, with DTX disabled. Tone signals were generated and processed in the range 50-3998 Hz with a frequency step of 21 Hz. Each tone lasted 8 seconds at a level of –26 dBovl. In order to discard potential transition

Joint SMG11/TSG-SA4 Meeting #13 / #8
Kyoto, Japan, 6-10 December 1999

Document S4-99454R
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

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06.93 CR A006Rev1 Current Version: 7.2.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to: TSG-SA#6
list expected approval meeting # here ↑

for approval	X	Strategic		*(for SMG use only)*
for information		non-strategic	X	

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects:
(at least one should be marked with an X)

(U)SIM	ME	UTRAN / Radio	Core Network
	X		X

Source: TSG-SA WG4 Codec **Date:** 16.12.1999

Subject: Editorial clarifications concerning RATSCCH and RX/TX DTX handler synchronisation at handover.

Work item: AMR

Category:
(only one category shall be marked with an X)

F	Correction	
A	Corresponds to a correction in an earlier release	
B	Addition of feature	
C	Functional modification of feature	
D	Editorial modification	X

Release:

Phase 2	
Release 96	
Release 97	
Release 98	X
Release 99	
Release 00	

Reason for change:
The block diagram displaying the transmit side DTX functions need to reflect the control signal flow from TX RSS to TX DTX handler.
Editorial clarification of TX DTX handler operation during handover.
Correction of editorial mistakes

Clauses affected: 5, 5.1.1, 5.1.2.1, 5.1.2.4, 6.1.1, 6.1.2

Other specs Affected:

Other 3G core specifications		→ List of CRs:
Other GSM core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

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5 Transmit (TX) side

A block diagram of the transmit side DTX functions is shown in figure 1.

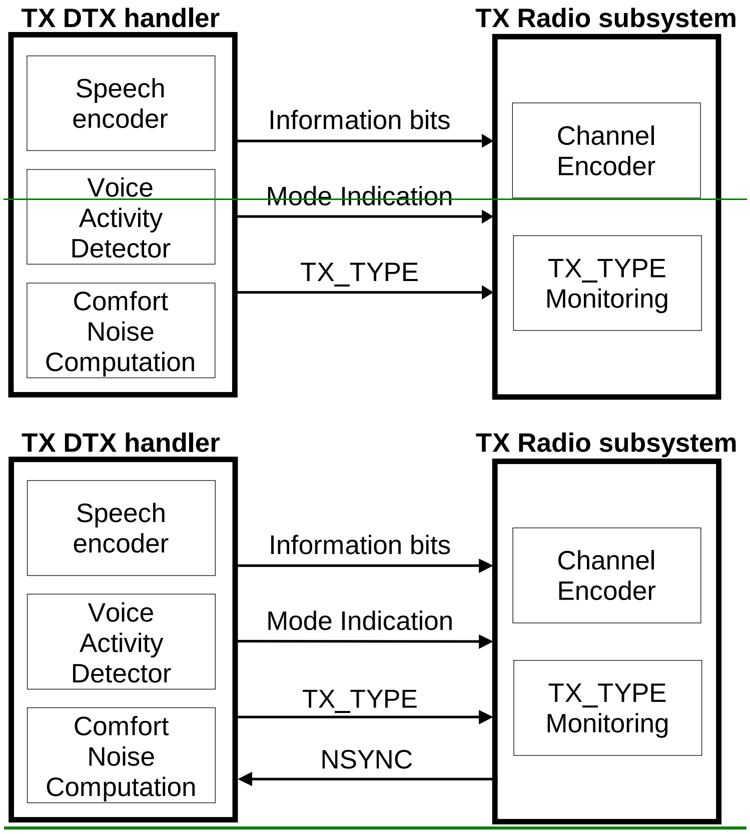

Figure 1: Block diagram of the transmit side DTX functions

5.1.1 Functions of the TX DTX handler

To allow an exact verification of the TX DTX handler functions, all frames before the reset of the system are treated as if there were speech frames of an infinitely long time. Therefore, and in order to ensure the correct estimation of comfort noise parameters at RX DTX side, the first 7 frames after the reset or after enabling the DTX operation shall always be marked with TX_TYPE= " SPEECH_GOOD ", even if VAD flag ="0" (hangover period, see figure 2).

The Voice Activity Detector (VAD) shall operate all the time in order to assess whether the input signal contains speech or not. The output is a binary flag (VAD flag ="1" or VAD flag ="0", respectively) on a frame by frame basis (see GSM 06.94).

The VAD flag controls indirectly, via the TX DTX handler operations described below, the overall DTX operation on the transmit side.

Whenever VAD flag ="1", the speech encoder output frame along with mode information shall be passed directly to the radio subsystem (RSS), marked with TX_TYPE =" SPEECH_GOOD "

At the end of a speech burst (transition VAD flag ="1" to VAD flag ="0"), it takes 8 consecutive frames to make a new updated SID analysis available at receiver side (see GSM 06.92). Normally, the first 7 speech encoder output frames after the end of the speech burst shall therefore be passed directly to the RSS, marked with TX_TYPE =" SPEECH_GOOD " ("hangover period"). The end of the speech is then indicated by passing frame 8 after the end of the speech burst to the RSS, marked with TX_TYPE = “SID_FIRST” (see figure 2).

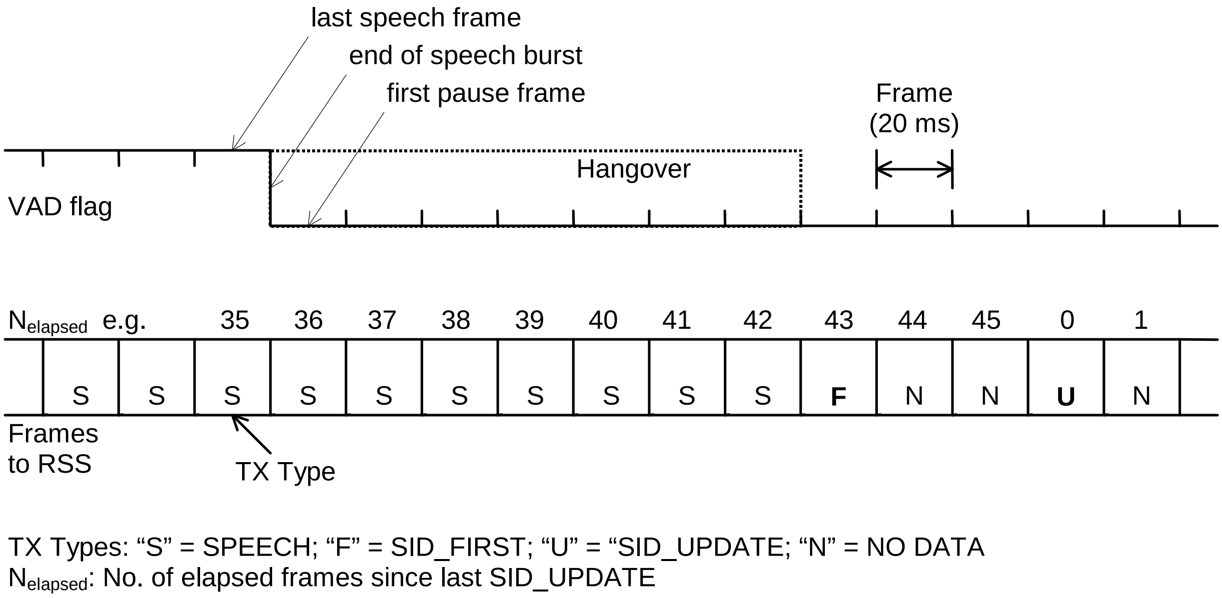

Figure 2: Normal hangover procedure ($N_{elapsed}$ > 23)

If, however, at the end of the speech burst, less than 24 frames have elapsed since the last SID_UPDATE frame was computed and passed to the RSS, then this last analysed SID_UPDATE frame shall repeatedly be passed to the RSS whenever a SID_UPDATE frame is to be produced, until a new updated SID analysis is available (8 consecutive frames marked with VAD flag ="0").This reduces the activity on the air in cases where short background noise spikes are taken for speech, by avoiding the "hangover" waiting for the SID frame computation.

Once the first SID analysis after the end of a speech burst has been computed and the SID_FIRST frame has been passed to the Radio Subsystem, the TX DTX handler shall at regular intervals compute and pass updated SID_UPDATE (Comfort Noise) frames to the Radio Subsystem (RSS) as long as VAD flag = "0". SID_UPDATE frames shall be generated every 8th frame. The first SID_UPDATE shall be sent as the third frame after the SID_FIRST frame.

The speech encoder is operated in full speech modality if TX_TYPE = " SPEECH_GOOD " and otherwise in a simplified mode, because not all encoder functions are required for the evaluation of comfort noise parameters and because comfort noise parameters are only to be generated at certain times.

In order to ensure TX/RX DTX handler synchronisation at handover, the uplink TX DTX handler in the MS shall accept messages from TX RSS with control parameter NSYNC, resulting in the following operation during the next NSYNC frames:

- The TX DTX handler shall send SID_UPDATE instead of NO_DATA frames to the TX RSS.
- Whenever~~If~~, during that period VAD flag = 1, ~~then~~ the TX DTX handler shall continue to produce SPEECH frames for at least the rest of the period and, in addition, the hangover period.

5.1.2.1 Functions of the TX Radio Subsystem for TCH/AFS

The TX Radio Subsystem operates in the following way regarding DTX (without TFO):

- all frames marked with TX_TYPE = " SPEECH_GOOD " are scheduled for normal channel coding and transmission. The frame format for CHE operation shall be SPEECH. If, however, the previous frame was not of TX_TYPE = "SPEECH_GOOD" or of TX_TYPE = "SID_FIRST", an ONSET frame format followed by SPEECH_GOOD shall be signalled to the CHE;

- for frames marked with TX_TYPE = "SID_FIRST" a SID_FIRST frame format is signalled to the CHE. Note: Normally, only the first 4 TDMA frames carrying bits of this frame (and carrying the second half of the preceding SPEECH frame) shall be transmitted. The remaining bits of the SID_FIRST frame pre-set the interleaver buffer with a special bit pattern for the case a frame marked with TX_TYPE = "SPEECH_GOOD" is immediately following;

frames marked with TX_TYPE = "SID_UPDATE" are scheduled for SID_UPDATE frame channel coding and transmission. The frame format signalled to CHE is SID_UPDATE;

for frames marked with TX_TYPE = "NO_DATA" no processing or transmission is carried out.

~~When operating in non-speech after a handover the GSM-MS shall schedule its most recent SID_UPDATE frame instead of the first incoming NO_DATA frame to update the remote speech decoder's Comfort Noise states. This action is aborted if the transcoder schedules a new SID_UPDATE frame.~~

If a SID_FIRST frame or the first SID_UPDATE frame after a SID_FIRST frame, is stolen for Fast Associated Control Channel (FACCH) signalling purposes, then the subsequent frame shall be scheduled for transmission of the SID_FIRST or SID_UPDATE frame (whichever applies) instead.

SPEECH frames shall override possible SID_FIRST or SID_UPDATE frames in exceptional cases.

At handover, TX/RX DTX handler synchronisation shall be initiated. At the time instant before the MS starts sending to the new base station, a message shall be sent to the uplink TX DTX handler with the parameter NSYNC = 12.

5.1.2.4 Functions of the TX Radio Subsystem for RATSCCH

During regular speech transmission (in the middle of a speech burst)RATSCCH replaces (steals) one (TCH/AFS) respectively two (TCH/AHS) speech frames (see GSM 05.09). Also in all non speech cases the RATSCCH shall be handled like speech. The respective RATSCCH frame formats (RATSCCH in case of TCH/AFS, respectively RATSCCH_MARKER and RATSCCH_DATA in case of TCH/AHS) shall be signalled to the CHE.

If RATSCCH has to be sent during a speech pause in DTX, then first an ONSET frame shall be signalled to the CHE, followed by the RATSCCH frame(s) and finally by the respective SID_FIRST frame(s).

If a SID_UPDATE frame is affected by RATSCCH signalling, then the SID_UPDATE frame shall be re-scheduled for transmission immediately after the RATSCCH signalling.

~~It is proposed (but not mandatory) to handle a~~ FACCH should be handled in the same way as a RATSCCH, i.e. like a short speech burst.

6.1.1 Functions of the RX radio subsystem

The RX radio subsystem uses a combination of gross-bit markers, receiver measurements, and CRC checks to classify each received frame. The basic operation for each frame is outlined below:

- the receiver first searches for the RATSCCH, SID_UPDATE, SID_FIRST or ONSET gross bit markers.

- If the RATSCCH signalling is detected, then the RATSCCH frame (TCH/AFS) respectively the RATSCCH_MARKER and RATSCCH_DATA frames (TCH/AHS) shall be decoded and handled as described in GSM 05.09. They shall be passed to the RX DTX handler as a NO_DATA frame(s).

- If the SID_FIRST marker is detected the frame is passed to the RX DTX handler as a SID_FIRST frame.

- If the SID_UPDATE marker is detected, then the frame ~~is~~ shall be decoded and passed to the RX DTX handler as a SID_UPDATE or a SID_BAD or a NO_DATA frame, depending on the CRC and the information bits, along with the comfort noise parameters, if applicable. A NO_DATA frame shall be passed on, if all information bits of a SID_UPDATE frame are set to “1” and the CRC is bad (see SID_FILLER in subclause 5.1.2.3).

- If the ONSET marker is detected, then an ONSET frame shall be passed to the RX DTX handler.

- if neither SID_UPDATE nor SID_FIRST markers are detected, the frame shall be channel decoded assuming it to be a speech frame. Depending on the CRC for speech frame channel decoding along with other receiver measurements the frame shall then be passed to the RX DTX handler marked as either SPEECH_GOOD, SPEECH_ DEGRADED, SPEECH_BAD or NO_DATA frame.

6.1.2 Functions of the RX DTX handler

The RX DTX handler is responsible for the overall DTX operation on the RX side. It consists of two main modes: SPEECH and COMFORT_NOISE. The initial mode shall be SPEECH.

The DTX operation on the RX side shall be as follows:

- The RX DTX handler shall enter mode SPEECH, when a frame classified as SPEECH_GOOD or SPEECH_DEGRADED is received. ONSET frames may be taken into account to identify the beginning of a speech burst;

whenever a frame classified as SPEECH_GOOD is received the RX DTX handler shall pass it directly on to the speech decoder;

- if the RX DTX handler is in mode SPEECH, then frames classified as SPEECH_DEGRADED, SPEECH_BAD or NO_DATA shall be substituted and muted as defined in GSM 06.91. Frames classified as NO_DATA shall be handled like SPEECH_BAD frames without valid speech information;

- frames classified as SID_FIRST, SID_UPDATE or SID_BAD shall bring the RX DTX handler into mode COMFORT_NOISE and shall result in comfort noise generation, as defined in GSM 06.92. SID_BAD frames shall be substituted and muted as defined in GSM 06.91. In mode COMFORT_NOISE the RX DTX handler shall ignore all unusable frames (NO_DATA, SPEECH_BAD) delivered by the RSS; comfort noise generation shall continue, until timeout may apply (see GSM 06.91);

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06.93	**CR**	**A007Rev1**	Current Version:	7.2.0

GSM (AA.BB) or 3G (AA.BBB) specification number ↑   *↑ CR number as allocated by MCC support team*

For submission to:	TSG-SA#6	for approval	X	strategic		*(for SMG use only)*
list expected approval meeting # here ↑		for information		non-strategic	X	

Form: CR cover sheet, version 2 for 3GPP and SMG   *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	X	UTRAN / Radio		Core Network	X

Source:	TSG-SA WG4 Codec	**Date:**	16.12.1999
Subject:	Onset frame signalling by the TX RSS.		
Work item:	AMR		

Category:	F	Correction	X	**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature			Release 97	
shall be marked	C	Functional modification of feature			Release 98	X
with an X)	D	Editorial modification			Release 99	
					Release 00	

Reason for change: Correction of an inconsistency: For TCH/AFS the ONSET marker should always be used if the present frame is speech and the preceding frame is not a speech frame. If the preceding frame is a SID_FIRST frame, currently, no ONSET frame is being signalled.

Clauses affected: 5.1.2

Other specs affected:	Other 3G core specifications		→ List of CRs:	
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

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5.1.2 Functions of the TX Radio Subsystem

The TX Radio Subsystem has the following overall functionality. The radio transmission is cut after the transmission of a SID_FIRST frame when the speaker stops talking. During speech pauses the transmission is resumed at regular intervals for transmission of one SID_UPDATE frame, in order to update the generated comfort noise on the RX side (and to improve the measurement of the link quality by the RSS). Note that the transcoder knows what frames to send. In the case when nothing is to be transmitted it outputs frames marked with TX_TYPE = ”NO_DATA”.

Within the TX Radio Subsystem the TX_TYPE Monitoring unit controls the operation of the Channel Encoder (as specified in GSM 05.03) and the Transmission of the frame. Control input to the TX_TYPE Monitoring unit is the TX_TYPE. Control output and input to the Channel Encoder are indicators specifying the frame format. These frame format indicators are defined in GSM 05.03, they are different for TCH/AFS and TCH/AHS.

5.1.2.1 Functions of the TX Radio Subsystem for TCH/AFS

The TX Radio Subsystem operates in the following way regarding DTX (without TFO):

- all frames marked with TX_TYPE = " SPEECH_GOOD " are scheduled for normal channel coding and transmission. The frame format for CHE operation shall be SPEECH. If, however, the previous frame was not of TX_TYPE = “SPEECH_GOOD”~~ or of TX_TYPE = “SID_FIRST”~~, an ONSET frame format followed by SPEECH_GOOD shall be signalled to the CHE;

- for frames marked with TX_TYPE = “SID_FIRST” a SID_FIRST frame format is signalled to the CHE~~. Note: Normally, only the first 4 TDMA frames carrying bits of this frame (and carrying the second half of the preceding SPEECH frame) shall be transmitted. The remaining bits of the SID_FIRST frame pre-set the interleaver buffer with a special bit pattern for the case a frame marked with TX_TYPE = “SPEECH_GOOD” is immediately following~~;

- frames marked with TX_TYPE = “SID_UPDATE” are scheduled for SID_UPDATE frame channel coding and transmission. The frame format signalled to CHE is SID_UPDATE;

- for frames marked with TX_TYPE = “NO_DATA” no processing or transmission is carried out.

When operating in non-speech after a handover the GSM-MS shall schedule its most recent SID_UPDATE frame instead of the first incoming NO_DATA frame to update the remote speech decoder’s Comfort Noise states. This action is aborted if the transcoder schedules a new SID_UPDATE frame.

SPEECH frames shall override possible SID_FIRST or SID_UPDATE frames in exceptional cases.

At handover, TX/RX DTX handler synchronisation shall be initiated. At the time instant before the MS starts sending to the new base station, a message shall be sent to the uplink TX DTX handler with the parameter NSYNC = 12.

3GPP/SMG Meeting #12 — ***Document* S4-99380**

St.-Laurent-du-Var, France, 18-22 Oct 1999

e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

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26.090 CR 001 — Current Version: 3.0.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ — *↑ CR number as allocated by MCC support team*

For submission to: TSG-SA#6	for approval	X	strategic		(for SMG
list expected approval meeting # here ↑	for information		non-strategic	X	use only)

Form: CR cover sheet, version 2 for 3GPP and SMG — *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: (U)SIM [ ] ME [X] UTRAN / Radio [ ] Core Network [X]
(at least one should be marked with an X)

Source: TSG-SA WG4 Codec — **Date:** 16/12/99

Subject: Bit allocation of the adaptive multi-rate codec

Work item: AMR

Category:
(only one category shall be marked with an X)

F	Correction	X	
A	Corresponds to a correction in an earlier release		
B	Addition of feature		
C	Functional modification of feature		
D	Editorial modification		

Release:

Phase 2	
Release 96	
Release 97	
Release 98	X
Release 99	
Release 00	

Reason for change: Bring bit order tables of encoder output parameters into line with the ANSI-C code for the GSM Adaptive Multi Rate speech codec.

Clauses affected: 7

Other specs affected:

Other 3G core specifications		→ List of CRs:
Other GSM core specifications		→ List of CRs:
MS test specifications		→ List of CRs:
BSS test specifications		→ List of CRs:
O&M specifications		→ List of CRs:

Other comments:

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7 Detailed bit allocation of the adaptive multi-rate codec

The detailed allocation of the bits in the adaptive multi-rate speech encoder is shown for each mode in table 9a-9h. These tables show the order of the bits produced by the speech encoder. Note that the most significant bit (MSB) of each codec parameter is always sent first.

Table 9a: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 244 bits/20 ms, 12.2 kbit/s mode

Bits (MSB-LSB)	Description
s1 - s7	index of 1st LSF submatrix
s8 - s15	index of 2nd LSF submatrix
s16 - s23	index of 3rd LSF submatrix
s24	sign of 3rd LSF submatrix
s25 - s32	index of 4th LSF submatrix
s33 - s38	index of 5th LSF submatrix
subframe 1	
s39 - s47	adaptive codebook index
s48 - s51	adaptive codebook gain
s52	sign information for 1st and 6th pulses
s53 - s55	position of 1st pulse
s56	sign information for 2nd and 7th pulses
s57 - s59	position of 2nd pulse
s60	sign information for 3rd and 8th pulses
s61 - s63	position of 3rd pulse
s64	sign information for 4th and 9th pulses
s65 - s67	position of 4th pulse
s68	sign information for 5th and 10th pulses
s69 - s71	position of 5th pulse
s72 - s74	position of 6th pulse
s75 - s77	position of 7th pulse
s78 - s80	position of 8th pulse
s81 - s83	position of 9th pulse
s84 - s86	position of 10th pulse
s87 - s91	fixed codebook gain
subframe 2	
s92 - s97	adaptive codebook index (relative)
s98 - s141	same description as s48 - s91
subframe 3	
s142 - s194	same description as s39 - s91
subframe 4	
s195 - s244	same description as s92 - s141

Table 9b: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 204 bits/20 ms, 10.2 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s17	index of 2nd LSF subvector
s18 – s26	index of 3rd LSF subvector
subframe 1	
s27 – s34	adaptive codebook index
s35	sign information for 1st and 5th pulses
s36	sign information for 2nd and 6th pulses
s37	sign information for ~~5th~~3rd and 7th pulses
s38	sign information for 4th and 8th pulses
s39-s48	position for 1st, 2nd, and 5th pulses
s49-s58	position for 3rd, 6th, and 7th pulses
s59-s65	position for 4th and ~~7~~8th pulses
s66 – s72	codebook gains
subframe 2	
s73 – s77	adaptive codebook index (relative)
s78 – s115	same description as s35 – s72
subframe 3	
s116 – s161	same description as s27 – s72
subframe 4	
s162 – s204	same description as s73 – s115

Table 9c: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 159 bits/20 ms, 7.95 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s9	index of 1st LSF subvector
s10 - s18	index of 2nd LSF subvector
s19 – s27	index of 3rd LSF subvector
subframe 1	
s28 – s35	adaptive codebook index
s36 – s~~38~~39	position of ~~1st~~4th pulse
s~~39~~40 – s~~41~~42	position of ~~2nd~~3rd pulse
s~~42~~43 – s~~44~~45	position of ~~3rd~~2nd pulse
s~~45~~46 – s48	position of ~~4th~~1st pulse
s49	sign information for ~~1st~~4th pulse
s50	sign information for ~~2nd~~3rd pulse
s51	sign information for ~~3rd~~2nd pulse
s52	sign information for ~~4th~~1st pulse
s53 – s56	adaptive codebook gain
s57 – s61	fixed codebook gain
subframe 2	
s62 – s67	adaptive codebook index (relative)
s68 – s93	same description as s36 – s61
subframe 3	
s94 – s127	same description as s28 – s61
subframe 4	
s128 – s159	same description as s62 – s93

Table 9d: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 148 bits/20 ms, 7.40 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s17	index of 2nd LSF subvector
s18 – s26	index of 3rd LSF subvector
subframe 1	
s27 – s34	adaptive codebook index
s35 – s~~37~~38	position of ~~1st~~4th pulse
s~~38~~39 – s~~40~~41	position of ~~2nd~~3rd pulse
s~~41~~42 - s~~43~~44	position of ~~3rd~~2nd pulse
s~~44~~45 – s47	position of ~~4th~~1st pulse
s48	sign information for ~~1st~~4th pulse
s49	sign information for ~~2nd~~3rd pulse
s50	sign information for ~~3rd~~2nd pulse
s51	sign information for ~~4th~~1st pulse
s52 – s58	codebook gains
subframe 2	
s59 – s63	adaptive codebook index (relative)
s64 – s87	same description as s35 – s58
subframe 3	
s88 – s119	same description as s27 – s58
subframe 4	
s120 – s148	same description as s59 – s87

Table 9e: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 134 bits/20 ms, 6.70 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s17	index of 2nd LSF subvector
s18 – s26	index of 3rd LSF subvector
subframe 1	
s27 – s34	adaptive codebook index
s35 – s~~37~~38	position of ~~1st~~3rd pulse
s~~38~~39 – s~~41~~42	position of 2nd pulse
s~~42~~43 – s45	position of ~~3rd~~1st pulse
s46	sign information for ~~1st~~3rd pulse
s47	sign information for 2nd pulse
s48	sign information for ~~3rd~~1st pulse
s49 – s55	codebook gains
subframe 2	
s56 – s59	adaptive codebook index (relative)
s60 – s80	same description as s35 – s55
subframe 3	
s81 – s109	same description as s27 – s55
subframe 4	
s110 – s134	same description as s56 – s80

Table 9f: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 118 bits/20 ms, 5.90 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s17	index of 2nd LSF subvector
s18 – s26	index of 3rd LSF subvector
subframe 1	
s27 – s34	adaptive codebook index
s35 – s~~38~~39	position of ~~1st~~2nd pulse
s~~39~~40 – s43	position of ~~2nd~~1st pulse
s44	sign information for ~~1st~~2nd pulse
s45	sign information for ~~2nd~~1st pulse
s46 – s51	codebook gains
subframe 2	
s52 – s55	adaptive codebook index (relative)
s56 – s72	same description as s35 – s51
subframe 3	
s73 – s97	same description as s27 – s51
subframe 4	
s98 – s118	same description as s52 – s72

Table 9g: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 103 bits/20 ms, 5.15 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s16	index of 2nd LSF subvector
s17 – s23	index of 3rd LSF subvector
subframe 1	
s24 – s31	adaptive codebook index
s32	position subset
s33 – s35	position of ~~1st~~2nd pulse
s36 – s38	position of ~~2nd~~1st pulse
s39	sign information for ~~1st~~2nd pulse
s40	sign information for ~~2nd~~1st pulse
s41 – s46	codebook gains
subframe 2	
s47 – s50	adaptive codebook index (relative)
s51 – s65	same description as s32 – s46
subframe 3	
s66 – s84	same description as s47 – s65
subframe 4	
s85 – s103	same description as s47 – s65

Table 9h: Source encoder output parameters in order of occurrence and bit allocation within the speech frame of 95 bits/20 ms, 4.75 kbit/s mode

Bits (MSB-LSB)	Description
s1 – s8	index of 1st LSF subvector
s9 - s16	index of 2nd LSF subvector
s17 – s23	index of 3rd LSF subvector
subframe 1	
s24 – s31	adaptive codebook index
s32	position subset
s33 – s35	position of ~~1st~~2nd pulse
s36 – s38	position of ~~2nd~~1st pulse
s39	sign information for ~~1st~~2nd pulse
s40	sign information for ~~2nd~~1st pulse
s41 – s48	codebook gains
subframe 2	
s49 – s52	adaptive codebook index (relative)
s53 – s61	same description as s32 – s40
subframe 3	
s62 - s65	same description as s49 – s52
s66 – s82	same description as s32– s48
subframe 4	
s83 – s95	same description as s49 – s61

3GPP/SMG Meeting #12 | ***Document*** ***S4-99392***

St.-Laurent-du-Var, France, 18-22 Oct 1999

e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

26.091	**CR**	**001**	Current Version:	3.0.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to:	TSG-SA#6	for approval	X	strategic		*(for SMG*
list expected approval meeting # here ↑		for information		non-strategic	X	*use only)*

Form: CR cover sheet, version 2 for 3GPP and SMG *The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc*

Proposed change affects: *(at least one should be marked with an X)*	(U)SIM		ME	X	UTRAN / Radio		Core Network	X

TSG-SA WG4 Codec	TSG-SA WG4 Codec	**Date:**	16/12/99
Subject:	Use of random excitation when RX_NODATA and not in DTX.		
Work item:	AMR		

Category:	F	Correction	X	**Release:**	Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature			Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	X
					Release 00	

Reason for change:	When the speech decoder receives no data, which results in zero speech parameters, a buzzy sound may be produced. This can be avoided by using pseudo random parameters instead of zero parameters. The speech decoder receives no data, e.g., due to the FACCH.
Clauses affected:	6.2.3.2 and 7.2.4

Other specs Affected:	Other 3G core specifications		→ List of CRs:	
	Other GSM core specifications	X	→ List of CRs:	06.73
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:	

help.doc

<--------- double-click here for help and instructions on how to create a CR.

6.2.3.2 Innovation sequence

The received fixed codebook innovation pulses from the erroneous frame are ~~always~~ used in the state in which they were received when corrupted data are received . In the case when no data were received random fixed codebook indicies should be employed.

7.2.4 Innovation sequence

The received fixed codebook innovation pulses from the erroneous frame are ~~always~~ used in the state in which they were received when corrupted data are received. In the case when no data were received random fixed codebook indicies should be employed.

7.3 Substitution and muting of lost SID frames

In the speech decoder a single frame classified as SID_BAD shall be substituted by the last valid SID frame information and the procedure for valid SID frames be applied. If the time between SID information updates (updates are specified by SID_UPDATE arrivals and ocassionally by SID_FIRST arrivals see 06.92) is greater than one second this shall lead to attenuation.

Joint SMG11/TSG-SA4 Meeting #13 / #8
Kyoto, Japan, 6-10 December 1999

Document S4-99538
e.g. for 3GPP use the format TP-99xxx
or for SMG, use the format P-99-xxx

CHANGE REQUEST

Please see embedded help file at the bottom of this page for instructions on how to fill in this form correctly.

26.093 CR 004rev2 Current Version: 3.0.1

GSM (AA.BB) or 3G (AA.BBB) specification number ↑ *↑ CR number as allocated by MCC support team*

For submission to: TSG-SA#6	for approval	X	strategic		*(for SMG use only)*
list expected approval meeting # here ↑	for information		non-strategic	X	

Form: CR cover sheet, version 2 for 3GPP and SMG   The latest version of this form is available from: ftp://ftp.3gpp.org/Information/CR-Form-v2.doc

Proposed change affects: (U)SIM [ ] ME [X] UTRAN / Radio [ ] Core Network [X]
(at least one should be marked with an X)

Source: TSG-SA **Date:** 16/12/99

Subject: Alignment to GSM 06.93

Work item: AMR

Category:				**Release:**		
	F	Correction	X		Phase 2	
	A	Corresponds to a correction in an earlier release			Release 96	
(only one category	B	Addition of feature			Release 97	
shall be marked	C	Functional modification of feature			Release 98	
with an X)	D	Editorial modification			Release 99	X
					Release 00	

Reason for change: Clarifications to main part of specification. Bring TX_TYPE/RX_TYPE definition inline with 26.101.
Make annex A identical to GSM 06.93 Release 98 (version 7.2.0) plus SMG11/TSG-SA4-approved CRs A006 and A007.

Clauses affected: Main part and annex A.

Other specs affected:				
	Other 3G core specifications		→ List of CRs:	
	Other GSM core specifications		→ List of CRs:	
	MS test specifications		→ List of CRs:	
	BSS test specifications		→ List of CRs:	
	O&M specifications		→ List of CRs:	

Other comments:

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1. Scope

This document describes the~~operation of the Adaptive Multi Rate speech codec during~~ Source Controlled Rate (SCR)~~operation.~~

~~For clarity, the~~operation of the Adaptive Multi-Rate speech Codec in Codec Types UMTS_AMR and GSM_AMR for the UMTS system. The implementation of this SCR operation is mandatory in all UMTS equipment.

The description is structured according to the block ~~diagrams in figures 1 and 3.This~~diagram in figure 1. This structure of distributing the various functions between system entities is not mandatory for implementation, as long as the operation on the speech decoder output remains the same.

~~The SCR functions described in this technical specification are mandatory for implementation in the UEs. The receiver requirements are mandatory for implementation in all Transcoders , the transmitter requirements only for those links where SCR will be used.~~

~~Annex A-E describes the interworking operation of AMR with GSM-EFR, TDMA-EFR, TDMA-US1 and PDC-EFR. This mode of operation is F.F.S.~~Annex A describes the Discontinuous Transmission (DTX) operation of the Adaptive Multi-Rate speech Codec in Codec Type GSM_AMR for the GSM system. This annex is the former GSM 06.93 (release 98).

Annexes B to E describe the SCR operation of the Adaptive Multi-Rate speech Codec in Codec Types GSM_EFR, TDMA_EFR, TDMA_US1 and PDC_EFR for the UMTS system.

2. Normative references

This document incorporates by dated and undated reference, provisions from other publications. These normative references are cited at the appropriate places in the text and the publications are listed hereafter. For dated references, subsequent amendments to or revisions of any of these publications apply to this document only when incorporated in it by amendment or revision. For undated references, the latest edition of the publication referred to applies.

[1] 3G TS 26.071 : "AMR Speech Codec; General description".

[2] 3G TS 26.073 : "AMR Speech Codec; ANSI-C code".

[3] 3G TS 26.074 : "AMR Speech Codec; Test sequences".

[4] 3G TS 26.090 : "AMR Speech Codec; Transcoding functions".

[5] 3G TS 26.091 : "AMR Speech Codec; Error concealment of lost frames".

[6] 3G TS 26.092 : "AMR Speech Codec; Comfort noise aspects".

[7] 3G TS 26.094 : "AMR Speech Codec; Voice Activity Detector (VAD)".

[8] 3G TS 26.101 : “AMR Speech Codec; Frame structure".

3. Definitions, symbols and abbreviations

3.1 Definitions

For the purpose of this document, the following definitions apply.

frame: Time interval of ~~20 msec.~~20 ms, corresponding to the time segmentation of the Adaptive ~~Multi-Rate speech transcoder~~Multi-Rate speech Codec ,also used as a short term for a traffic frame.

traffic frame: Block of 95..244 information bits transmitted on the speech traffic channels.

SID frame: Frame that conveys information about the acoustic background noise.

speech frame: Traffic frame that has been classified as ~~a~~ SPEECH_GOOD or SPEECH_BAD frame.

VAD flag: Boolean flag, generated by the VAD algorithm indicating the presence ("1") or the absence ("0") of a speech frame.

RX_TYPE: ~~, generated by the de-framing unit , indicating to the RX SCR handler the type of data in the current frame. Refer to Table 2 for an example.~~

~~**TX_TYPE:** flag, generated by the TX SCR handler, indicating to the framin unit the type of data in the current frame. Refer to Table 1 for an example.~~classifies the received frame.

TX_TYPE: classifies the frame to be transmitted.

hangover period: A period of frames added at the end of a speech burst in which VAD flag ="0" and TX_TYPE is ~~="00",~~="SPEECH_GOOD", this period provides the encoder with an extra window to ~~analyze~~derive the Comfort Noise parameters .

3.2 Symbols

For the purpose of this document, the following symbols apply.

$N_{elapsed}$	Number of elapsed frames since the last updated SID frame.

3.3 Abbreviations

For the purpose of this document , the following abbreviations apply.

AN	Access Network
SCR	Source Controlled Rate operation
TS	Telecommunication Standard, Technical Specification
GSM	Global System for Mobile Telecommunication~~s~~
GSM-EFR	GSM Enhanced Full Rate speech ~~codec~~Codec
UE	User Equipment
PDC-EFR	ARIB PDC-EFR 6.7 ~~kbit/s speech coder~~kBit/s speech Codec
RAN	Radio ~~AN~~Access Network
RX	Receive
SID	Silence Descriptor ~~(Background character Descriptor)~~
TDMA-EFR	TIA IS-641 Enhanced speech ~~coder~~Codec
TDMA-US1	TIA TDMA-US1 (12.2 ~~kbit/s EFR)~~kBit/s Codec, similar to GSM-EFR)
TX	Transmit
VAD	Voice Activity Detector

4. General

Source Controlled Rate operation (SCR) is a mechanism ~~in~~for the AMR Speech Codec, which allows~~the codec~~ to encode ~~speech~~the input signal at a lower average rate by taking speech inactivity into account. The SCR scheme may be used for the following purposes:

- to save power in the User Equipment;
- to reduce the overall interference and load in the networks.

SCR in the transmitting path (uplink) shall be in operation in UEs, if commanded so by the network. The UE shall handle SCR in the receiving path (downlink) at any time, regardless, whether SCR in the transmitting path is commanded or not.

4.1 General organisation

The default SCR mechanism described in this document requires the following functions:

- —a Voice Activity Detector (VAD) on the transmit (TX) side;
- —evaluation of the background acoustic noise on the transmit (TX) side, in order to transmit characteristic parameters to the receive (RX) side;
- —generation on the receive (RX) side of a similar noise, called comfort noise, during periods where the transmission is switched off.

The Voice Activity Detector (VAD) is defined in [7] and the AMR-mode comfort noise functions in [6]. Both are based partly on the speech ~~transcoder~~Codec and its internal variables, defined in [4].

In addition to these functions, if the parameters arriving at the RX side are detected to be seriously corrupted by errors, the speech or comfort noise must be generated from substituted data in order to avoid seriously annoying effects for the listener. These functions~~for the AMR-mode~~ are defined in [5].

An overall description of the speech processing parts can be found in [1]. An overview of one link SCR operation is shown in Figure 1.

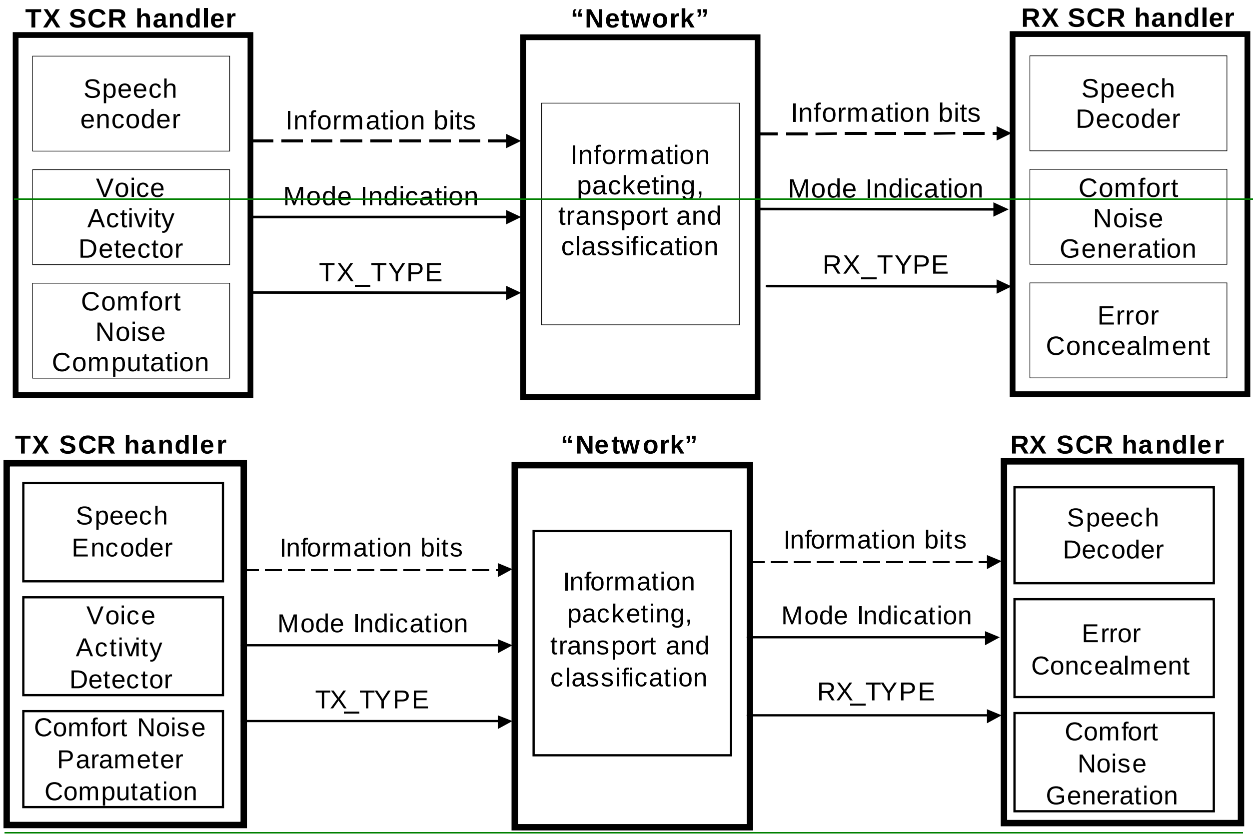


Figure 1: Block diagram of one link SCR operation

5. AMR SCR operation

5.1 Transmit (TX) side

A block diagram of the transmit side SCR functions is shown in Figure 2.

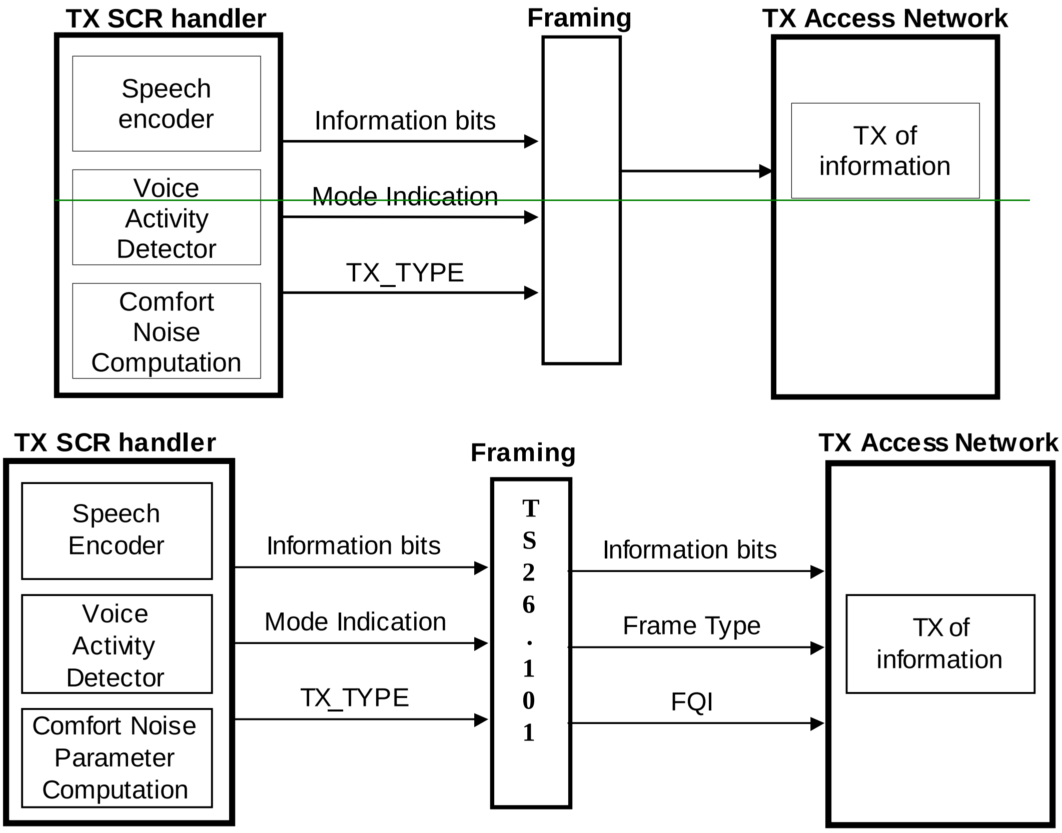


Figure 2: Block diagram of ~~the transmit side SCR functions~~SCR functions at the TX side

5.1.1 General operation

The TX SCR handler passes traffic frames, individually marked by TX_TYPE, to the Framing unit. Each frame consists of bit fields containing the information bits, the codec mode indication, and the TX_TYPE. TX_TYPE shall be used to specify the contents of the frame. The table below provides an overview of the different TX_TYPEs used and explains the required contents in the information bit and the mode indication bit fields.

~~TX_TYPE~~	~~Legend~~	~~Information Bits~~	~~Mode Indication~~
~~00~~	~~SPEECH~~	~~speech frame, size depending on codec mode~~	~~current codec mode~~
~~01~~	~~SID_FIRST~~ ~~(end of speech marker, start of CN generation)~~	~~no useful information,~~	~~the codec mode that would have been used if TX_TYPE had been 00 (SPEECH)~~
~~10~~	~~SID_UPDATE~~	~~comfort noise information , information bits~~	~~the codec mode that would have been used if TX_TYPE had been 00 (SPEECH)~~
~~11~~	~~NO_DATA~~	~~no useful information~~	~~no useful information~~

TX_TYPE	Information Bits	Mode Indication
SPEECH_GOOD	Speech frame, size 95..244 bits, depending on codec mode	Current codec mode
SPEECH_BAD	Corrupt speech frame (bad CRC), size 95..244 bits, depending on codec mode	Current codec mode
SID_FIRST	Marker for the end of talkspurt, no further information, all 35 comfort noise bits set to "0"	The codec mode that would have been used if TX_TYPE had been "SPEECH_GOOD"
SID_UPDATE	35 comfort noise bits	The codec mode that would have been used if TX_TYPE had been "SPEECH_GOOD"
SID_BAD	Corrupt SID update frame (bad CRC)	The codec mode that would have been used if TX_TYPE had been "SPEECH_GOOD"
NO_DATA	No useful information, nothing to be transmitted	No useful information

Table 1: ~~TX-TYPE~~SCR TX_TYPE identifiers for UMTS_AMR and GSM_AMR

TX_TYPE = ~~"11"~~"NO_DATA" indicates that the Information Bit and Codec Mode fields do not contain any useful data (and should not be transmitted over AN). The purpose of this TX_TYPE is to provide the option to save network transmission between the transcoder and AN. Note, the TX_TYPEs "SPEECH_BAD" and "SID_BAD" may occur in TFO and TrFO situations.

The scheduling of the frames for transmission on the Access Network is controlled by the TX SCR handler by the use of the TX_TYPE field~~ and the given SCR operation mode~~.

5.1.2 ~~5.1.2~~ Functions of the TX SCR handler

If TX SCR operation is disabled, the TX SCR handler continuously generates speech frames, i.e. frames marked with TX_TYPE="SPEECH_GOOD".

If the TX SCR operation is enabled, the VAD flag controls the TX SCR handler operation as described in the following paragraphs.

5.1.2.1 AMR SCR Timing procedures

To allow an exact verification of the TX SCR handler functions, all frames before the reset of the system are treated as if there were speech frames of an infinitely long time. Therefore, and in order to ensure the correct estimation of comfort noise parameters at RX SCR side, the first ~~seven~~7 frames after the reset ~~are~~or after enabling the SCR operation shall always be marked with TX_TYPE= ~~"00"”~~,"SPEECH_GOOD", even if VAD flag ="0" (hangover period, see figure 3).

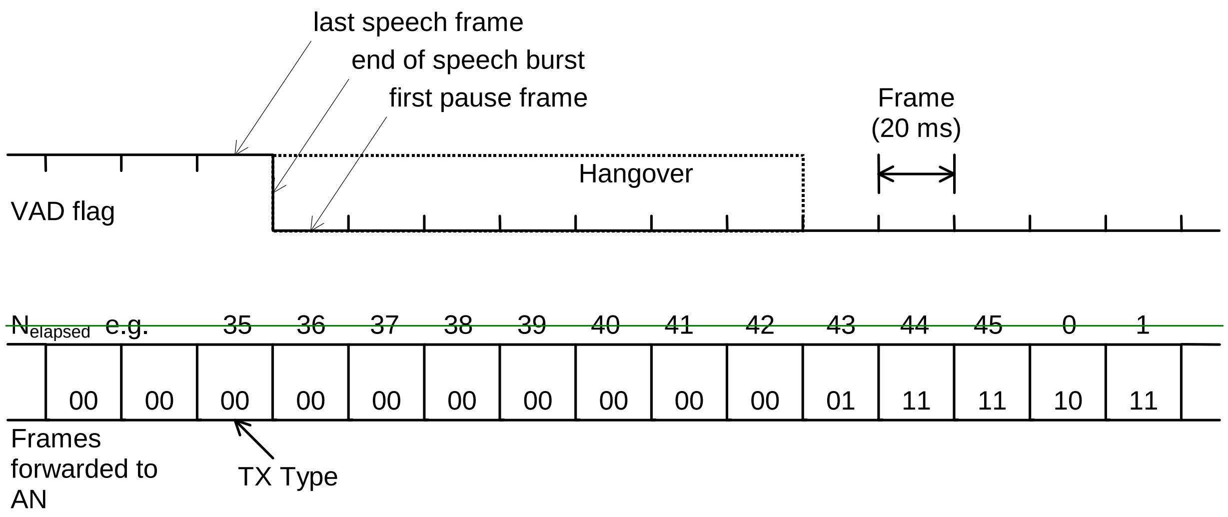
The Voice Activity Detector (VAD) shall operate all the time in order to assess whether the input signal contains speech or not. The output is a binary flag (VAD flag ="1" or VAD flag ="0", respectively) on a frame by frame basis (see [7]).

The VAD flag controls indirectly, via the TX SCR handler operations described below, the overall SCR operation on the transmit side.

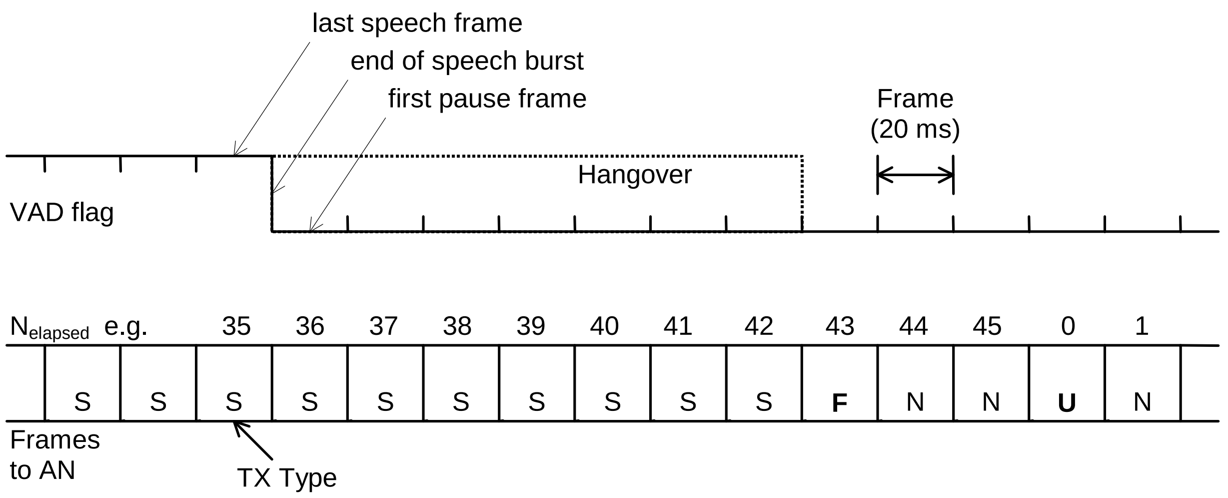
Whenever VAD flag ="1", the speech encoder output frame along with mode information shall be passed directly to the AN~~)~~, marked with TX_TYPE ~~="00"~~=" SPEECH_GOOD "

At the end of a speech burst (transition VAD flag ="1" to VAD flag ="0"), it takes eight consecutive frames to make a new updated SID analysis available (see [6]). Normally, the first seven speech encoder output frames after the end of the speech burst shall therefore be passed directly to the AN, marked with TX_TYPE ~~="00" (SPEECH)~~= “SPEECH_GOOD” ("hangover period").

The end of the speech is then indicated by passing frame eight after the end of the speech burst to the AN, marked with TX_TYPE = ~~“01” (SID_FIRST) (see figure 2).~~“SID_FIRST” (see figure 3). SID_FIRST frames do not contain data.



TX Types: “00” = SPEECH; “01” = SID_FIRST; “10” = “SID_UPDATE; “11” = NO DATA
$N_{elapsed}$: No. of elapsed frames since last SID_UPDATE



TX Types: “S” = SPEECH; “F” = SID_FIRST; “U” = “SID_UPDATE; “N” = NO DATA
$N_{elapsed}$: No. of elapsed frames since last SID_UPDATE

Figure 3: Normal hangover procedure for AMR ($N_{elapsed}$ > 23)

If, however, at the end of the speech burst, less than 24 frames have elapsed since the last SID_UPDATE frame was computed, then this last analysed SID_UPDATE frame should be passed to the AN whenever a SID_UPDATE frame~~(TX_TYPE=“10”)~~ is to be produced, until a new updated SID analysis is available (8 consecutive frames marked with VAD flag ="0"). This reduces the load on the network in cases where short background noise spikes are taken for speech, by avoiding the "hangover" waiting for the SID frame computation.

Once the ~~first SID analysis after the end of a speech burst has been computed and the first SID~~SID_FIRST frame has been passed to the AN, the TX SCR handler shall at regular intervals compute and pass updated SID_UPDATE (Comfort Noise) frames~~(TX_TYPE = “10”)~~ to the AN~~)~~ as long as VAD flag = "0". SID_UPDATE frames shall be generated every 8th frame. ~~However for frame loss robustness the~~The first SID_UPDATE~~frame~~ shall be sent as the third frame after the~~initial~~ SID_FIRST frame.

~~Note:~~ The speech encoder is operated in full speech modality if TX_TYPE = ~~"00"~~" SPEECH_GOOD" and otherwise in a simplified mode, because not all encoder functions are required for the evaluation of comfort noise parameters and because comfort noise parameters are only to be generated at certain times.

5.1.3 The TX part of the AN

The TX part of the AN has the following overall functionality. The transmission is cut after the transmission of a SID_FIRST frame when the speaker stops talking. During speech pauses the transmission is resumed at regular intervals for transmission of one SID_UPDATE frame, in order to update the generated comfort noise on the RX side. The~~transcoder decides what frames to send. In the case whennothing is to be transmitted it outputs frames marked with TX_TYPE = ”11”.~~

~~Demands on~~operation of the TX part of the ~~Access Network~~

AN is controlled ~~The TX part of the AN operates in the following way regarding SCR:~~

~~- frames marked with TX_TYPE = “00” (SPEECH) are scheduled for transmission.~~

~~- frames marked with TX_TYPE = “01” (SID_FIRST) are scheduled for transmission.~~

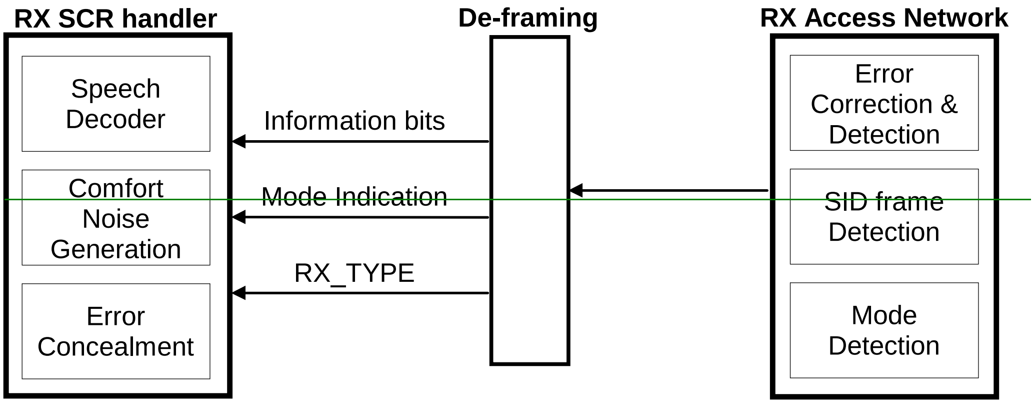
~~- frames marked with TX_TYPE = “10” (SID_UPDATE) are scheduled for transmission~~

~~- for frames marked with TX_TYPE = “11” (NO_DATA) no processing or transmission is carried out.~~by the TX SCR handler via the TX_TYPE.

~~SPEECH frames shall override possible~~All frames, marked with SPEECH_GOOD, SID_FIRST or SID_UPDATE ~~frames in these exceptional cases.~~ shall be transmitted by the TX part of the AN.

5.2 Receive (RX) side

A block diagram of the receive side SCR functions is shown in Figure 3 below.



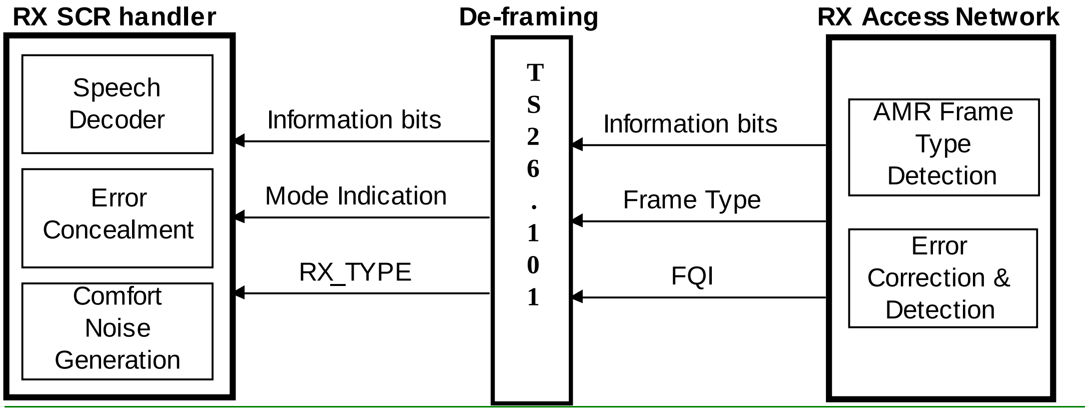


Figure 4: Block diagram of the receive side SCR functions

5.2.1 General operation

~~Whatever their context (speech, SID, or none), the deframing unit after~~The AN~~continuously~~ passes ~~the~~all received traffic frames to the RX SCR handler, ~~individually marked by various pre-processing functions with a 3 bit type indicator RX_TYPE~~classified with RX_TYPE, as described in Table ~~2, which serve to classify the traffic frame. This classification allows the RX SCR handler to determine in a simple way how the received~~ 2 (see TS 26.102). The RX SCR handles the frame accordingly.

~~frame is to be handled.~~

~~RX_TYPE~~	~~Legend~~	~~Description~~
~~000~~	~~SPEECH_GOOD~~	~~Speech frame without errors in class A, and most likely not in class B (quality information in the RX part of the AN also OK)~~
~~001~~	~~SPEECH_PROBABLY_DEGRADED~~	~~Speech frame with CRC OK (class A OK), but less sensitive bits may be corrupted. At least 1 bit error in the speech coder's class B bits is suspected~~

~~010~~	~~**SPEECH_BAD**~~	~~(likely) speech frame with bad CRC (or estimated to be very bad by the RX part of the AN)~~
~~011~~	~~**SPARE**~~	~~Spare~~
~~100~~	~~**SID_FIRST**~~	~~This SID-frame marks the beginning of a comfort noise period.~~
~~101~~	~~**SID_UPDATE**~~	~~Correct SID update frame~~
~~110~~	~~**SID_BAD**~~	~~Corrupt SID update frame (bad CRC; applicable only for SID_UPDATE frames)~~
~~111~~	~~**NO_DATA**~~	~~Nothing useable was received. The synthesis mode of the previous frame type is used.~~

RX_TYPE	**Information Bits**
SPEECH_GOOD	Speech frame without detected errors.
SPEECH_BAD	(likely) speech frame with bad CRC (or estimated to be very bad by the RX part of the AN)
SID_FIRST	This SID-frame marks the beginning of a comfort noise period.
SID_UPDATE	Correct SID update frame
SID_BAD	Corrupt SID update frame (bad CRC; applicable only for SID_UPDATE frames)
NO_DATA	Nothing useable was received. The synthesis mode of the previous frame type is used.

Table 2: RX_TYPE identifiers for UMTS_AMR and GSM_AMR

~~5.2.2 RX part of the AN~~

~~The RX/de-framing unit uses a combination of measurements from AN , and CRC checks to classify each received frame according to RX_TYPE (Table 2).~~

5.2.3 ~~5.2.3~~ Demands on the RX SCR handler

The RX SCR handler is responsible for the overall SCR operation on the RX side. It consists of two main modes: SPEECH and COMFORT_NOISE. The initial mode shall be SPEECH.

The SCR operation on the RX side shall be as follows:

- ~~whenever~~The RX SCR handler shall enter mode SPEECH, when a frame classified as SPEECH_GOOD is received.
- whenever a frame classified as SPEECH_GOOD is received the RX SCR handler shall pass it directly on to the speech decoder;
- ~~when a frame classified as SPEECH_PROBABLY_DEGRADED, SPEECH_BAD, SID_BAD is received the error concealment procedure(s) shall be applied. If not in Comfort Noise Generation mode, the RX SCR handler shall treat NO_DATA frames delivered by the AN as~~if the RX SCR handler is in mode SPEECH, then

frames classified as SPEECH_BAD or NO_DATA shall be substituted and muted as defined in [5]. Frames classified as NO_DATA shall be handled like SPEECH_BAD frames without valid speech ~~information.;~~ ~~-~~information;

- ~~- frames classified as SID_UPDATE, SID_FIRST or SID_BAD shall result in Comfort Noise Generation mode, until the next SID_UPDATE frame has arrived or~~ frames classified as ~~SPEECH_OK or SPEECH_PROBABLY_DEGRADED are detected. During this period,~~SID_FIRST, SID_UPDATE or SID_BAD shall bring the RX SCR handler into mode COMFORT_NOISE and shall result in comfort noise generation, as defined in [6]. SID_BAD frames shall be substituted and muted as defined in [5];

in mode COMFORT_NOISE the RX SCR handler shall ignore ~~any unusable frames (NO_DATA) delivered by the AN;~~

all unusable frames (NO_DATA, SPEECH_BAD); comfort noise generation ~~- Frames classified as SPARE are handled as NO_DATA frames.~~

- ~~5.3 AMR SID Information format~~shall continue, until timeout may apply ([5]);

Annex A: AMR DTX handler for the GSM system (corresponding to GSM 06.93)

A.1 Scope

The present document gives a description of the general baseband operation of Adaptive Multi-Rate speech traffic channels in the transmitter and in the receiver of GSM Mobile Stations (MS)s and Base Station Systems (BSS)s during Discontinuous Transmission (DTX).

For clarity, the description is structured according to the block diagrams in figures 1 and 3. Except in the case described next, this structure of distributing the various functions between system entities is not mandatory for implementation, as long as the operation on the air interface and on the speech decoder output remains the same.

In the case of BSSs where the speech transcoder is located remote from the Base Transceiver Station (BTS), the implementation of the interfaces between the DTX handlers and the Radio Sub System (RSS) as described in the present document together with all their flags is mandatory, being part of the A-bis interface as described in GSM 08.60 and GSM 08.61.

The DTX functions described in this technical specification are mandatory for implementation in the GSM MSs. The receiver requirements are mandatory for implementation in all GSM BSSs, the transmitter requirements only for those where downlink DTX or Tandem Free Operation will be used.

A.2 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.
- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.
- For this Release 1999 document, references to GSM documents are for Release 1999 versions (version 7.x.y).

[1] 3G TS 21.004: "Digital cellular telecommunication system (Phase 2+); Abbreviations and acronyms".

[2] 3G TS 24.008: "Digital cellular telecommunication system (Phase 2+); Mobile radio interface layer 3 specification".

[3] 3G TS 25.003: "Digital cellular telecommunication system (Phase 2+); Channel coding".

[4] 3G TS 25.005: "Digital cellular telecommunication system (Phase 2+); Radio transmission and reception".

[5] 3G TS 25.008: "Digital cellular telecommunication system (Phase 2+); Radio subsystem link control".

[6] 3G TS 25.009: "Digital cellular telecommunication system (Phase 2+); Link adaptation".

[7] 3G TS 26.071: "Digital cellular telecommunications system (Phase 2+); Adaptive Multi-Rate (AMR) speech processing functions; General description".

[8] 3G TS 26.073: "Digital cellular telecommunications system (Phase 2+); ANSI-C code for the GSM Adaptive Multi-Rate speech codec".

[9] 3G TS 26.074: "Digital cellular telecommunications system (Phase 2); Test vectors for the GSM Adaptive Multi-Rate speech codec".

[10] 3G TS 26.090: "Digital cellular telecommunications system (Phase 2+); Adaptive Multi-Rate speech transcoding".

[11] 3G TS 26.091: "Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frame for Adaptive Multi-Rate speech traffic channels".

[12] 3G TS 26.092: "Digital cellular telecommunications system (Phase 2+); Comfort noise aspects for Adaptive Multi-Rate speech traffic channels".

[13] 3G TS 26.094: "Digital cellular telecommunications system (Phase 2+); Voice Activity Detector (VAD) for Adaptive Multi-Rate speech traffic channels".

[14] 3G TS 28.060: "Digital cellular telecommunication system (Phase 2+); Inband control of remote transcoders and rate adaptors for Full Rate traffic channels".

[15] 3G TS 28.061: "Digital cellular telecommunication system (Phase 2+); Inband Control of Remote Transcoders and Rate Adaptors for Half Rate traffic channels".

[16] 3G TS 28.062: " Digital cellular telecommunications system; Inband Tandem Free Operation (TFO) of Speech Codecs".

A.3 Definitions, symbols and abbreviations

A.3.1 Definitions

For the purpose of the present document, the following definitions apply.

frame: Time interval of 20 ms, corresponding to the time segmentation of the Adaptive Multi Rate speech transcoder (3G TS 26.090 [9]), also used as a short term for a traffic frame.

traffic frame: Block of 95..244 information bits transmitted on the TCH/AFS or TCH/AHS speech traffic channels.

SID frame: Frame characterised by the SID (Silence Descriptor) gross bit patterns. It may convey information on the acoustic background noise.

speech frame: Traffic frame that has been classified as a SPEECH frame.

VAD flag: Boolean flag, generated by the VAD algorithm defined in 3G TS 26.094 indicating the presence ("1") or the absence ("0") of a speech frame.

RX_TYPE: flag with eight values, generated by the RX radio subsystem, indicating to the RX DTX handler the type of data in the current frame. Refer to Table 2.

TX_TYPE: flag with eight values, generated by the TX DTX handler, indicating to the TX radio subsystem the type of data in the current frame. Refer to Table 1.

hangover period: A period of 7 frames added at the end of a speech burst in which VAD flag ="0" and TX_TYPE is "SPEECH".

A.3.2 Symbols

For the purpose of the present document, the following symbols apply.

$N_{elapsed}$ Number of elapsed frames since the last updated SID frame.

A.3.3 Abbreviations

For the purpose of the present document, the following abbreviations apply.

BSC	Base Station Controller
BSS	Base Station System
BTS	Base Transceiver Station
CHD	Channel Decoder
CHE	Channel Encoder
DTX	Discontinuous Transmission
ETS	European Telecommunication Standard
FACCH	Fast Associated Control CHannel
GSM	Global System for Mobile Telecommunications
MS	Mobile Station
RATSCCH	Robust Amr Traffic Synchronised Control CHannel
RSS	Radio Sub System
RX	Receive
SACCH	Slow Associated Control CHannel
SID	SIlence Descriptor
TX	Transmit
VAD	Voice Activity Detector

For abbreviations not given in this subclause, see 3G TS 21.004.

A.4 General

Discontinuous Transmission (DTX) is a mechanism, which allows the radio transmitter to be switched off most of the time during speech pauses for the following two purposes:

- to save power in the Mobile Station (MS);
- to reduce the overall interference level over the air interface.

DTX in uplink shall be in operation within the GSM MS, if commanded so by the network, see 3G TS 204.08. The MS shall handle DTX in downlink at any time, regardless, whether DTX in uplink is commanded or not.

A.4.1 General organisation

The overall DTX mechanism described in the present document requires the following functions:

- a Voice Activity Detector (VAD) on the transmit (TX) side;
- evaluation of the background acoustic noise on the transmit (TX) side, in order to transmit characteristic parameters to the receive (RX) side;
- generation on the receive (RX) side of a similar noise, called comfort noise, during periods where the radio transmission is switched off.

The Voice Activity Detector (VAD) is defined in 3G TS 26.094 and the comfort noise functions in 3G TS 26.092. Both are based partly on the speech transcoder and its internal variables, defined in 3G TS 26.090.

In addition to these functions, if the parameters arriving at the RX side are detected to be seriously corrupted by errors, the speech or comfort noise must be generated from substituted data in order to avoid seriously annoying effects for the listener. This function is defined in 3G TS 26.091.

An overall description of the speech processing parts can be found in 3G TS 26.071.

The description for Tandem Free Operation is given in 3G TS 28.062.

A.5 Transmit (TX) side

A block diagram of the transmit side DTX functions is shown in figure 1.

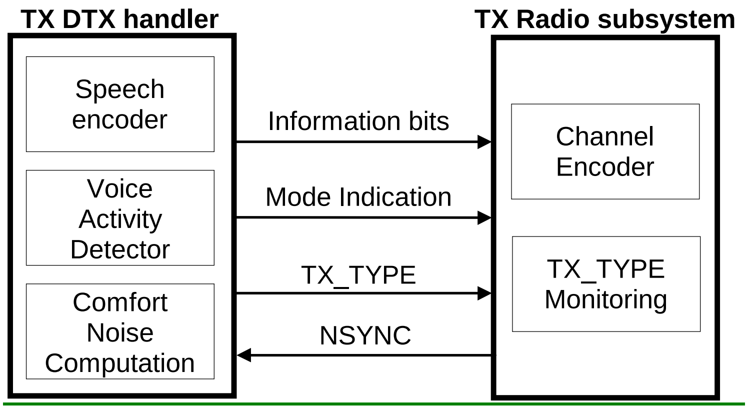


Figure 5: Block diagram of the transmit side DTX functions

A.5.1 General operation

The TX DTX handler passes traffic frames, individually marked by TX_TYPE, to the Radio Subsystem (RSS). Each frame passed to the RSS consists of bit fields containing the information bits, the codec mode indication, and the TX_TYPE. TX_TYPE is used to specify the contents of the frame. The table below provides an overview of the different TX_TYPEs used and explains the required contents in the information bit and the mode indication bit fields. In case of ongoing Tandem Free Operation (see 3G TS 28.062) frames with errors may arrive in downlink in the BTS.

Table 3: TX TYPE identifiers

TX_TYPE Legend	Information Bits	Mode Indication
SPEECH_GOOD	speech frame, size 95..244 bits depending on codec mode; no errors known.	current code mode
SPEECH_DEGRADED (only in downlink in TFO)	Speech frame, size 95..244 bits, depending on codec mode; there might be errors in class 2 bits.	current codec mode
SPEECH_BAD (only in downlink in TFO)	Speech frame, size 95..244 bits, depending on codec mode; there are errors in class 1 bits.	current codec mode
SID_FIRST	marks the end of a talkspurt, respectively the beginning of a speech pause; does not contain information bits.	the codec mode that would have been used if TX_TYPE had been SPEECH
SID_UPDATE	comfort noise, 35 bits; no errors known	the codec mode that would have been used if TX_TYPE had been SPEECH
SID_BAD (only in downlink in TFO)	comfort noise, 35 bits; errors detected, parameters unusable	the codec mode that would have been used if TX_TYPE had been SPEECH
ONSET (only in downlink in TFO)	announces the beginning of a speech burst; does not contain information bits	the codec mode of the following speech frame
NO_DATA	no useful information	no useful information

TX_TYPE = "NO_DATA" indicates that the Information Bit and Codec Mode fields do not contain any useful data (and shall not be transmitted over the air interface). The purpose of this TX_TYPE is to provide the option to save transmission between the transcoder and the radio base station if a packet oriented transmission is used.

The scheduling of the frames for transmission on the air interface is controlled by the TX DTX handler by the use of the TX_TYPE field.

A.5.1.1 Functions of the TX DTX handler

To allow an exact verification of the TX DTX handler functions, all frames before the reset of the system are treated as if there were speech frames of an infinitely long time. Therefore, and in order to ensure the correct estimation of comfort noise parameters at RX DTX side, the first 7 frames after the reset or after enabling the DTX operation shall always be marked with TX_TYPE= " SPEECH_GOOD ", even if VAD flag ="0" (hangover period, see figure 2).

The Voice Activity Detector (VAD) shall operate all the time in order to assess whether the input signal contains speech or not. The output is a binary flag (VAD flag ="1" or VAD flag ="0", respectively) on a frame by frame basis (see 3G TS 26.094).

The VAD flag controls indirectly, via the TX DTX handler operations described below, the overall DTX operation on the transmit side.

Whenever VAD flag ="1", the speech encoder output frame along with mode information shall be passed directly to the radio subsystem (RSS), marked with TX_TYPE =" SPEECH_GOOD "

At the end of a speech burst (transition VAD flag ="1" to VAD flag ="0"), it takes 8 consecutive frames to make a new updated SID analysis available at receiver side (see 3G TS 26.092). Normally, the first 7 speech encoder output frames after the end of the speech burst shall therefore be passed directly to the RSS, marked with TX_TYPE =" SPEECH_GOOD " ("hangover period"). The end of the speech is then indicated by passing frame 8 after the end of the speech burst to the RSS, marked with TX_TYPE = “SID_FIRST” (see figure 2).

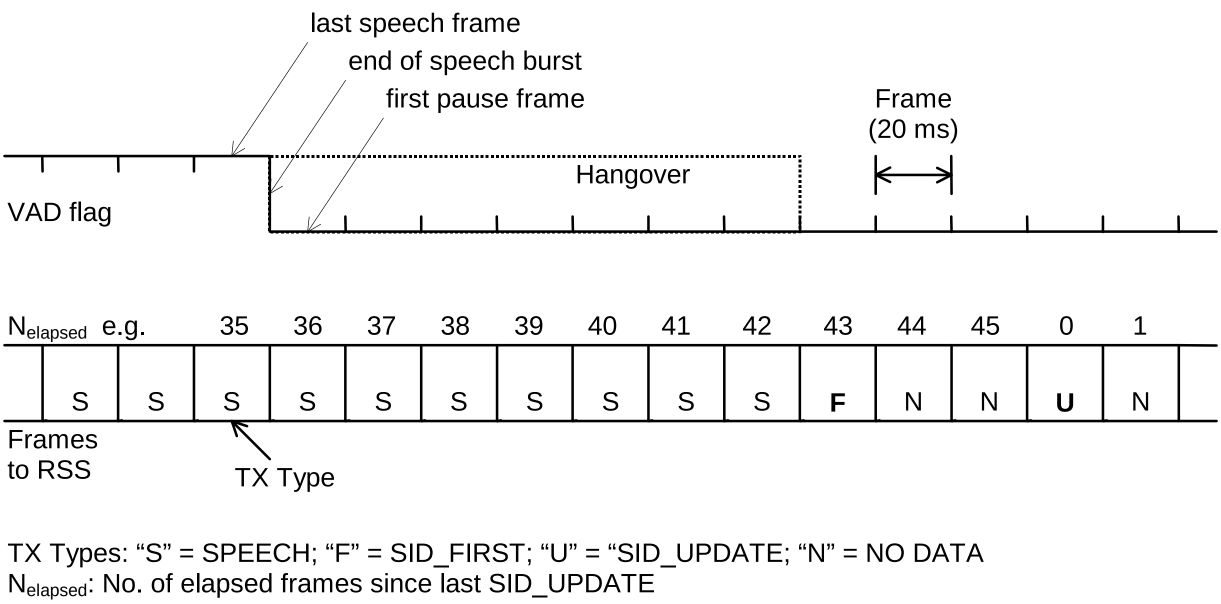


Figure 6: Normal hangover procedure ($N_{elapsed}$ > 23)

If, however, at the end of the speech burst, less than 24 frames have elapsed since the last SID_UPDATE frame was computed and passed to the RSS, then this last analysed SID_UPDATE frame shall repeatedly be passed to the RSS whenever a SID_UPDATE frame is to be produced, until a new updated SID analysis is available (8 consecutive frames marked with VAD flag ="0").This reduces the activity on the air in cases where short background noise spikes are taken for speech, by avoiding the "hangover" waiting for the SID frame computation.

Once the first SID analysis after the end of a speech burst has been computed and the SID_FIRST frame has been passed to the Radio Subsystem, the TX DTX handler shall at regular intervals compute and pass updated SID_UPDATE (Comfort Noise) frames to the Radio Subsystem (RSS) as long as VAD flag = "0". SID_UPDATE frames shall be generated every 8th frame. The first SID_UPDATE shall be sent as the third frame after the SID_FIRST frame.

The speech encoder is operated in full speech modality if TX_TYPE = " SPEECH_GOOD " and otherwise in a simplified mode, because not all encoder functions are required for the evaluation of comfort noise parameters and because comfort noise parameters are only to be generated at certain times.

In order to ensure TX/RX DTX handler synchronisation at handover, the uplink TX DTX handler in the MS shall accept messages from TX RSS with control parameter NSYNC, resulting in the following operation during the next NSYNC frames:

- The TX DTX handler shall send SID_UPDATE instead of NO_DATA frames to the TX RSS.
- Whenever, during that period VAD flag = 1, the TX DTX handler shall continue to produce SPEECH frames for at least the rest of the period and, in addition, the hangover period.

A.5.1.2 Functions of the TX Radio Subsystem

The TX Radio Subsystem has the following overall functionality. The radio transmission is cut after the transmission of a SID_FIRST frame when the speaker stops talking. During speech pauses the transmission is resumed at regular intervals for transmission of one SID_UPDATE frame, in order to update the generated comfort noise on the RX side (and to improve the measurement of the link quality by the RSS). Note that the transcoder knows what frames to send. In the case when nothing is to be transmitted it outputs frames marked with TX_TYPE = "NO_DATA".

Within the TX Radio Subsystem the TX_TYPE Monitoring unit controls the operation of the Channel Encoder (as specified in 3G TS 25.003) and the Transmission of the frame. Control input to the TX_TYPE Monitoring unit is the TX_TYPE. Control output and input to the Channel Encoder are indicators specifying the frame format. These frame format indicators are defined in 3G TS 25.003, they are different for TCH/AFS and TCH/AHS.

A.5.1.2.1 Functions of the TX Radio Subsystem for TCH/AFS

The TX Radio Subsystem operates in the following way regarding DTX (without TFO):

- all frames marked with TX_TYPE = " SPEECH_GOOD " are scheduled for normal channel coding and transmission. The frame format for CHE operation shall be SPEECH. If, however, the previous frame was not of TX_TYPE = "SPEECH_GOOD", an ONSET frame format followed by SPEECH_GOOD shall be signalled to the CHE;
- for frames marked with TX_TYPE = "SID_FIRST" a SID_FIRST frame format is signalled to the CHE;
- frames marked with TX_TYPE = "SID_UPDATE" are scheduled for SID_UPDATE frame channel coding and transmission. The frame format signalled to CHE is SID_UPDATE;
- for frames marked with TX_TYPE = "NO_DATA" no processing or transmission is carried out.

If a SID_FIRST frame or the first SID_UPDATE frame after a SID_FIRST frame, is stolen for Fast Associated Control Channel (FACCH) signalling purposes, then the subsequent frame shall be scheduled for transmission of the SID_FIRST or SID_UPDATE frame (whichever applies) instead.

SPEECH frames shall override possible SID_FIRST or SID_UPDATE frames in exceptional cases.

At handover, TX/RX DTX handler synchronisation shall be initiated. At the time instant before the MS starts sending to the new base station, a message shall be sent to the uplink TX DTX handler with the parameter NSYNC = 12.

A.5.1.2.2 Functions of the TX Radio Subsystem for TCH/AHS

The TX Radio Subsystem operates in the following way regarding DTX:

- all frames marked with TX_TYPE = " SPEECH_GOOD " are scheduled for normal channel coding and transmission. The frame format for CHE operation shall be SPEECH. However, if the previous frame was of TX_TYPE = "SID_FIRST", a SID_FIRST_INH frame format followed by SPEECH_GOOD shall be signalled to the CHE. If the previous frame was of TX_TYPE = "SID_UPDATE", a SID_UPDATE_INH frame format followed by SPEECH_GOOD shall be signalled to the CHE. If the previous frame was of TX_TYPE "NO_DATA", an ONSET frame format followed by SPEECH_GOOD shall be signalled to the CHE;

- for frames marked with TX_TYPE = “SID_FIRST” a SID_FIRST_P1 frame format is signalled to the CHE. Note: All 4 TDMA frames carrying the bits of this frame shall be transmitted. The Mode Indication received with the frame is stored for potential use in the next frame;

- ~~When the TX SCR handler is ordered by the network to operate in AMR mode with SCR operation enabled the~~for frames marked with TX_TYPE = “SID_UPDATE” a SID_UPDATE frame format is ~~according to [5]. This is the default and only mandatory operating mode of the SCR handler.~~

signalled to the CHE. All 4 TDMA frames carrying the bits of this frame shall be transmitted;

~~Annex A:~~
~~ETSI-AMR SCR handler~~

- ~~The ETSI-AMR SCR handler is indentical to the AMR SCR handler as described in the main body of this specification. Speech coding interworking aspects ETSI-AMR are described in [TBD].~~for frames marked with TX_TYPE = “NO_DATA”, no processing or transmission is carried out. However, if the preceding frame was marked with TX_TYPE = “SID_FIRST”, a SID_FIRST_P2 frame format is signalled to CHE. Note: The 2 TDMA frames carrying bits of this frame shall be transmitted. If, depending on the current frame number, the Mode Indication is to be transmitted with these TDMA frames, the Mode Indication shall be used that was stored during the processing of the preceding SID_FIRST frame.

If a SID_FIRST frame or the first SID_UPDATE frame after a SID_FIRST frame, is affected by Fast Associated Control Channel (FACCH) signalling purposes, then the SID_FIRST or SID_UPDATE frame (whichever applies) shall be re-scheduled for transmission immediately after the FACCH signalling.

SPEECH frames shall override possible SID_FIRST or SID_UPDATE frames in exceptional cases.

At handover, TX/RX DTX handler synchronisation shall be initiated. At the time instant before the MS starts sending to the new base station, a message shall be sent to the uplink TX DTX handler with the parameter NSYNC = 12.

A.5.1.2.3 Functions of the Downlink TX Radio Subsystem for TFO

The TX Radio Subsystem in the BTS shall in addition operate in the following way regarding DTX, if TFO is ongoing (see 3G TS 28.062):

- Frames with TX_TYPE = SPEECH_GOOD, SID_FIRST and SID_UPDATE shall be handled as usual in DTX, regardless whether DTX in downlink is requested or not. Also NO_DATA shall be handled as usual, if DTX is requested.
- Frames with TX_TYPE = NO_DATA shall be replaced by SID_FILLER frames, if DTX in downlink is not requested. By this the radio transmission continues in downlink, although no parameters are transmitted in speech pauses on the Abis interface. The MS generates Comfort Noise in these speech pauses.
- Frames with TX_TYPE = SPEECH_DEGRADED shall be handled exactly like SPEECH_GOOD frames.
- For frame with TX_TYPE = SPEECH_BAD and SID_BAD the CHE shall perform its regular processing, but then shall invert the six, respectively 14 CRC bits before convolutional encoding and transmitting the frames on the air interface. By this the error concealment mechanism in the MS is triggered to handle these corrupted frames.
- ONSET frames may be ignored by the TX Radio Subsystem and need not to be processed.

Definition: SID_FILLER frames are like SID_BAD frames, but with all information bits set to “1”. The 14 CRC bits shall artificially be inverted by the CHE before convolutional encoding and transmission.

A.5.1.2.4 Functions of the TX Radio Subsystem for RATSCCH

During regular speech transmission (in the middle of a speech burst)RATSCCH replaces (steals) one (TCH/AFS) respectively two (TCH/AHS) speech frames (see 3G TS 25.009). Also in all non speech cases the RATSCCH shall be handled like speech. The respective RATSCCH frame formats (RATSCCH in case of TCH/AFS, respectively RATSCCH_MARKER and RATSCCH_DATA in case of TCH/AHS) shall be signalled to the CHE.

If RATSCCH has to be sent during a speech pause in DTX, then first an ONSET frame shall be signalled to the CHE, followed by the RATSCCH frame(s) and finally by the respective SID_FIRST frame(s).

If a SID_UPDATE frame is affected by RATSCCH signalling, then the SID_UPDATE frame shall be re-scheduled for transmission immediately after the RATSCCH signalling.

FACCH should be handled in the same way as a RATSCCH, i.e. like a short speech burst.

A.6 Receive (RX) side

A block diagram of the receive side DTX functions is shown in figure 3.

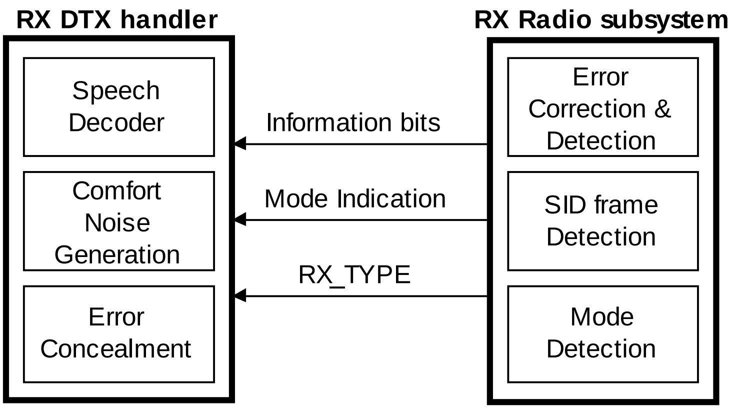


Figure 7: Block diagram of the receive side DTX functions

A.6.1 General operation

Whatever their context (speech, SID, FACCH or none), the RSS continuously passes the received traffic frames to the RX DTX handler, individually marked by various pre-processing functions with RX_TYPE as described in subclause 6.1.1 and table 2, which serves to classify the traffic frame. This classification allows the RX DTX handler to determine in a simple way how the received frame is to be handled.

Table 4: RX_TYPE identifiers

RX_TYPE Legend	Description
SPEECH_GOOD	Speech frame with CRC OK, Channel Decoder soft values also OK
SPEECH_DEGRADED	Speech frame with CRC OK, but 1B bits and class2 bits may be corrupted
SPEECH_BAD	(likely) speech frame, bad CRC (or very bad Channel Decoder measures)
SID_FIRST	first SID marks the beginning of a comfort noise period
SID_UPDATE	SID update frame (with correct CRC)
SID_BAD	Corrupt SID update frame (bad CRC; applicable only for SID_UPDATE frames)
ONSET	ONSET frames precede the first speech frame of a speech burst
NO_DATA	Nothing useable (for the speech decoder) was received. This applies for the cases of no received frames (DTX) or received FACCH or RATSCCH or SID_FILLER signalling frames.

A.6.1.1 Functions of the RX radio subsystem

The RX radio subsystem uses a combination of gross-bit markers, receiver measurements, and CRC checks to classify each received frame. The basic operation for each frame is outlined below:

- the receiver first searches for the RATSCCH, SID_UPDATE, SID_FIRST or ONSET gross bit markers.

- If the RATSCCH signalling is detected, then the RATSCCH frame (TCH/AFS) respectively the RATSCCH_MARKER and RATSCCH_DATA frames (TCH/AHS) shall be decoded and handled as described in 3G TS 25.009. They shall be passed to the RX DTX handler as a NO_DATA frame(s).
- If the SID_FIRST marker is detected the frame is passed to the RX DTX handler as a SID_FIRST frame.
- If the SID_UPDATE marker is detected, then the frame shall be decoded and passed to the RX DTX handler as a SID_UPDATE or a SID_BAD or a NO_DATA frame, depending on the CRC and the information bits, along with the comfort noise parameters, if applicable. A NO_DATA frame shall be passed on, if all information bits of a SID_UPDATE frame are set to "1" and the CRC is bad (see SID_FILLER in subclause 5.1.2.3).
- If the ONSET marker is detected, then an ONSET frame shall be passed to the RX DTX handler.
- if neither SID_UPDATE nor SID_FIRST markers are detected, the frame shall be channel decoded assuming it to be a speech frame. Depending on the CRC for speech frame channel decoding along with other receiver measurements the frame shall then be passed to the RX DTX handler marked as either SPEECH_GOOD, SPEECH_ DEGRADED, SPEECH_BAD or NO_DATA frame.

A.6.1.2 Functions of the RX DTX handler

The RX DTX handler is responsible for the overall DTX operation on the RX side. It consists of two main modes: SPEECH and COMFORT_NOISE. The initial mode shall be SPEECH.

The DTX operation on the RX side shall be as follows:

- The RX DTX handler shall enter mode SPEECH, when a frame classified as SPEECH_GOOD or SPEECH_DEGRADED is received. ONSET frames may be taken into account to identify the beginning of a speech burst;
- whenever a frame classified as SPEECH_GOOD is received the RX DTX handler shall pass it directly on to the speech decoder;
- if the RX DTX handler is in mode SPEECH, then frames classified as SPEECH_DEGRADED, SPEECH_BAD or NO_DATA shall be substituted and muted as defined in 3G TS 26.091. Frames classified as NO_DATA shall be handled like SPEECH_BAD frames without valid speech information;

frames classified as SID_FIRST, SID_UPDATE or SID_BAD shall bring the RX DTX handler into mode COMFORT_NOISE and shall result in comfort noise generation, as defined in 3G TS 26.092. SID_BAD frames shall be substituted and muted as defined in 3G TS 26.091. In mode COMFORT_NOISE the RX DTX handler shall ignore all unusable frames (NO_DATA, SPEECH_BAD) delivered by the RSS; comfort noise generation shall continue, until timeout may apply (see 3G TS 26.091);