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TSG-S4#6 approved the technical content of the 3G TS 26.073 Version 1.0.0 presented for information to TSG-SA#5.

TSG-S4 understands that this document, which contains a copy of the ANSI C-Code of the Adaptive Multi-Rate speech codec, will be made available on the free-access 3GPP Web site, after its presentation to TSG-SA.

Understanding that the 3G specifications do not contain any Copyright Statement, TSG-S4 would like to remind TSG-SA that part of the code included in this specification is also included in specifications from different sources (for example GSM EFR or AMR code in ETSI specifications). In these specifications the code is protected by a Copyright Statement.

Since S4 is not competent to discuss copyright issues, we would like to point out this difference for TSG-SA to consider when reviewing this specification.

TS 26.073 V1.0.0 (1999-10)

Technical Specification

**3rd Generation Partnership Project (3GPP)
TSG-SA Codec Working Group
ANSI-C code for the Adaptive Multi Rate speech codec**



3GPP

Reference

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Intellectual Property Rights

Foreword

This Technical Specification has been produced by the 3rd Generation Partnership Project, Technical Specification Group Services and System Aspects, Working Group 4 (Codec).

The contents of this informal TS may be subject to continuing work within the 3GPP and may change following formal TSG-S4 approval. Should TSG-S4 modify the contents of this TS, it will be re-released with an identifying change of release date and an increase in version number as follows:

- Version m.t.e
- where:
 - m indicates [major version number]
 - x the second digit is incremented for all changes of substance, i.e. technical enhancements, corrections, updates, etc.
 - y the third digit is incremented when editorial only changes have been incorporated into the specification.

1 Scope

This Technical Standard (TS) contains an electronic copy of the ANSI-C code for the Adaptive Multi-Rate codec. The ANSI-C code is necessary for a bit exact implementation of the Adaptive Multi Rate speech transcoder (TS 26.090 [2]), Voice Activity Detection (TS 26.094 [6]), comfort noise (TS 26.092 [4]), source controlled rate operation (TS 26.093 [5]) and example solutions for substituting and muting of lost frames (TS 26.091 [3]).

2 Normative references

This TS 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 TS only when incorporated in it by amendment or revision. For undated references, the latest edition of the publication referred to applies.

- [1] TS 26.074 : "AMR Speech Codec; Test sequences".
- [2] TS 26.090 : "AMR Speech Codec; Speech transcoding".
- [3] TS 26.091 : "AMR Speech Codec; Substitution and muting of lost frames".
- [4] TS 26.092 : "AMR Speech Codec; Comfort noise aspects".
- [5] TS 26.093 : "AMR Speech Codec; Source controlled rate operation".
- [6] TS 26.094 : "AMR Speech Codec; Voice Activity Detection"

3 Definitions and abbreviations

3.1 Definitions

Definition of terms used in the present document, can be found in TS 06.090 [2], TS 06.091 [3], TS 06.092 [4], TS 06.093 [5] and TS 06.094 [6].

3.2 Abbreviations

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

- ANSI American National Standards Institute
- ETS European Telecommunication Standard
- GSM Global System for Mobile communications
- I/O Input/Output
- RAM Random Access Memory
- ROM Read Only Memory

4 C code structure

This clause gives an overview of the structure of the bit-exact C code and provides an overview of the contents and organization of the C code attached to this document.

The C code has been verified on the following systems:

Sun Microsystems workstations and GNU gcc compiler;

DEC Alpha workstations and GNU gcc compiler;

IBM PC/AT compatible computers with Linux operating system and GNU gcc compiler;

ANSI-C 9899 was selected as the programming language because portability was desirable

4.1 Contents of the C source code

The C code distribution has all files in the root level.

The distributed files with suffix ".c" contain the source code and the files with suffix ".h" are the header files. The ROM data is contained mostly in files with suffix ".tab".

The C code distribution also contains one speech coder installation verification data file, "spch_dos.inp". The reference encoder output file is named "spch_dos.cod", the reference decoder input file is named "spch_dos.dec" and the reference decoder output file is named "spch_dos.out". These four files are formatted such that they are correct for an

IBM PC/AT compatible computer. The same files with reversed byte order of the 16 bit words are named "spch_unx.inp", "spch_unx.cod", "spch_unx.dec" and "spch_unx.out", respectively. Final verification is to be performed using the Adaptive Multi-Rate test sequences described in TS 06.074 [1]. Makefiles are provided for the platforms in which the C code has been verified (listed above). Once the software is installed, this directory will have a compiled version of encoder and decoder (the bit-exact C executables of the speech codec) and all the object files.

4.2 Program execution

The Adaptive Multi-Rate codec is implemented in two programs:

- (*encoder*) speech encoder;
- (*decoder*) speech decoder.

The programs should be called like:

```
encoder [encoder options] <speech input file> <parameter file>
```

```
decoder [decoder options] <parameter file> <speech output file>
```

The speech files contain 16-bit linear encoded PCM speech samples and the parameter files contain encoded speech data and some additional flags.

The encoder and decoder options will be explained by running the applications with option `-h`. See the file `readme.txt` for more information on how to run the *encoder* and *decoder* programs.

4.3 Coding style

The C code is written according to the following structuring conventions. Each function `func()` that needs static variables is considered a module. A module consists of:

- a 'state structure' (struct) combining the static variables of the module
- three auxiliary functions `func_init()`, `func_reset()`, and `func_exit()`.
- the processing function `func()` itself

The initialization function `func_init()` allocates (from the heap) a new state structure, calls the `func_reset()` function, stores the pointer to the newly allocated structure in its first function parameter, and returns with a value of 0 if completed successful or a value of 1 otherwise.

The reset function `func_reset()` takes a pointer to the state structure and resets all members of the structure to a predefined value ('homing').

The exit function `func_exit()` performs any necessary cleanup and frees the state structure memory.

The processing function `func()` also takes a pointer to the state structure as well as all other necessary parameters and performs its task using (and possibly modifying) the values in the state structure.

If a module calls other modules, the higher level state structure contains a pointer to the lower level state structures, and the `init`, `reset`, and `exit` functions recursively call the corresponding lower level functions.

By this convention, the code becomes "instantiable" (more than one copy of a module can be used in the same program) and the static data hierarchy is clearly visible in the code.

4.4 Code hierarchy

Figures 1 to 4 are call graphs that show the functions used in the speech codec, including the functions of VAD, DTX, and comfort noise generation.

Each column represents a call level and each cell a function. The functions contain calls to the functions in rightwards neighboring cells. The time order in the call graphs is from the top downwards as the processing of a frame advances. All standard C functions: `printf()`, `fwrite()`, etc. have been omitted. Also, no basic operations (`add()`, `L_add()`, `mac()`, etc.) or double precision extended operations (e.g. `L_Extract()`) appear in the graphs. The initialization of the static RAM (i.e. calling the `_init` functions) is also omitted.

The basic operations are not counted as extending the depth, therefore the deepest level in this software is level 7.

The encoder call graph is broken down into three separate call graphs, Table 1 to 3.

Table 1: Speech encoder call structure

Speech_Encode_Frame	Pre_Process					
	cod_amr	Copy				
		Vad1 ¹	filter_bank	first_filter_stage		
				filter5		
				filter3		
				level_calculation		
			vad_decision	complex_estimate_adapt		
				complex_vad		
				noise_estimate_update	update_cntrl	
				hangover_addition		
			Vad2 ¹	block_norm		
				r_fft	c_fft	
				fn10Log10	Log2	Log2_norm
				Pow2		
			tx_dtx_handler			
			lpc	Autocorr		
				Lag_window		
				Levinson		
			lsp	Az_lsp	Chebps	
				Q_plsf_5	Lsp_lsf	
					Lsf_wt	
					Vq_subvec	
					Vq_subvec_s	
					Reorder_lsf	
					Lsf_lsp	
				Int_lpc_1and3_2	Lsp_az	Get_lsp_pol
				Int_lpc_1and3	Lsp_az	Get_lsp_pol
				Q_plsf_3	Lsp_lsf	
					Lsf_wt	
					Copy	
					Vq_subvec3	
					Vq_subvec4	
					Reorder_lsf	
					Lsf_lsp	
				Int_lpc_1to3_2	Lsp_az	Get_lsp_pol
				Int_lpc_1to3	Lsp_az	Get_lsp_pol
				Copy		
			dtx_buffer	Copy		
				Log2	Log2_norm	
			dtx_enc	Lsp_lsf		
				Reorder_lsf		
				Lsf_lsp		
			Set_zero			
			lsp_reset	Copy		
				Q_plsf_reset		
			cl_ltp_reset	Pitch_fr_reset		
	check_lsp					
	pre_big	Weight_Ai				
		Residu				
		Syn_filt				
	ol_ltp	Pitch_ol	vad_tone_detection_update ²			
			Lag_max	vad_tone_detection ²		
				Inv_sqrt		
			comp_corr ²			
			hp_max ²			
			vad_complex_detection_update ²			
		Pitch_ol_wgh	comp_corr ²			
			Lag_max ²	vad_tone_detection_update ²		
				vad_tone_detection ²		
			gmed_n			
			hp_max ²			
			vad_complex_detection_update ²			
	vad_pitch_detection	LTP_flag_update ³				
	subframePreProc	Weight_Ai				
		Syn_filt				
		Residu				
		Copy				
	cl_ltp	Pitch_fr	getRange			
			Norm_Corr	Convolve		
				Inv_sqrt		
			searchFrac	Interpol_3or6		
			Enc_lag3			
			Enc_lag6			

(continued)

1 Option to call one or the other VAD option
 2 Specific to VAD option 1
 3 Specific to VAD option 2

Table 1 (concluded): Speech encoder call structure

		Pred_lt_3or6
		Convolve
		G_pitch
		check_gp_clipping
		q_gain_pitch
	cbsearch	see Table 2
	gainQuant	see Table 3
	update_gp_clipping	Copy
	subframePostProc	Syn_filt
		Pred_lt_3or6
		Convolve
	Prm2bits	Int2bin

Table 2: cbsearch call structure

cbsearch	code_2i40_9bits	cor_h_x	
		set_sign	
		cor_h	Inv_sqrt
		search_2i40	
		build_code	
		cor_h_x	
	code_2i40_11bits	set_sign	
		cor_h	Inv_sqrt
		search_2i40	
		build_code	
		cor_h_x	
		set_sign	
	code_3i40_14bits	cor_h	Inv_sqrt
		search_3i40	
		build_code	
	code_4i40_17bits	cor_h_x	
		set_sign	
		cor_h	Inv_sqrt
		search_4i40	
		build_code	
		cor_h_x	
	code_8i40_31bits	set_sign12k2	Inv_sqrt
		cor_h	Inv_sqrt
		search_10and8i40	
	build_code		
	compress_code	compress10	
	cor_h_x		
code_10i40_35bits	set_sign12k2	Inv_sqrt	
	cor_h	Inv_sqrt	
	search_10and8i40		
	build_code		
	q_p		

Table 3: gainQuant call structure

gainQuant	gc_pred_copy	Copy		
	gc_pred	Log2	Log2_norm	
		Log2_norm		
	calc_filt_energies			
	calc_target_energy			
	MR475_update_unq_pred	gc_pred_update		
	MR475_gain_quant	MR475_quant_store_results	Log2	Log2_norm
		gc_pred	gc_pred_update	Log2_norm
			Log2_norm	
	G_code			
	q_gain_code	Pow2		
	MR795_gain_quant	q_gain_pitch		
		MR795_gain_code_quant3		
		calc_unfilt_energies	Log2	Log2_norm
		gain_adapt	gmed_n	
		MR795_gain_code_quant_mod	sqrt_l_exp	
Qua_gain	Pow2			
gc_pred_update				

Table 4: Speech decoder call structure

Speech_Decode_Frame	Bits2prm	Bin2int				
	Decoder_amr	rx_dtx_handler				
		Decoder_amr_reset		lsp_avg_reset		
				D_plsf_reset		
				ec_gain_pitch_reset		
				ec_gain_code_reset		
				gc_pred_reset		
				Bgn_scd_reset	Set_zero	
				ph_disp_reset		
				dtx_dec_reset	Copy	
					Set_zero	
			dtx_dec		Copy	
				Lsf_lsp		
				Init_D_plsf_3	Copy	
				D_plsf_3	Reorder_lsf	
					Copy	
					Lsf_lsp	
				pseudonoise		
				Lsp_lsf		
				Reorder_lsf		
				Lsp_Az	Get_lsp_pol	
				A_Refl		
				Log2	Log2_norm	
				Build_CN_code	pseudonoise	
				Syn_filt		
				Lsf_lsp		
			Lsp_avg			
			Copy			
			D_plsf_3	Reorder_lsf		
				Copy		
				Lsf_lsp		
			Int_lpc_1to3	Lsp_Az	Get_lsp_pol	
			D_plsf_5	Reorder_lsf		
				Copy		
				Lsf_lsp		
			Int_lpc_1and3	Lsp_Az	Get_lsp_pol	
			Dec_lag3			
			Pred_lt_3or6			
			Dec_lag6			
			decode_2i40_9bits			
			decode_2i40_11bits			
			decode_3i40_14bits			
			decode_4i40_17bits			
			decode_8i40_31bits	decompress_code	decompress10	
			ec_gain_pitch	gmed_n		
			d_gain_pitch			
			ec_gain_pitch_update			
			decode_10i40_35bits			
			Dec_gain	Log2	Log2_norm	
				gc_pred	Log2	Log2_norm
				Log2_norm		
				Pow2		
				gc_pred_update		
			ec_gain_code	gmed_n		
				gc_pred_average_limited		
				gc_pred_update		
			ec_gain_code_update			
			d_gain_code	gc_pred	Log2	Log2_norm
					Log2_norm	
				Pow2		
				gc_pred_update		
		Int_lsf				
		Cb_gain_average				
	ph_disp_release					
	ph_disp_lock					
	ph_disp					
	sqrt_l_exp					
	Ex_ctrl	gmed_n				
	agc2	Inv_sqrt				
	Syn_filt					
	Bgn_scd	gmed_n				
	dtx_dec_activity_update	Copy				
		Log2	Log2_norm			
	Lsp_avg					
Post_Filter		Copy				
		Weight_Ai				
		Residu				
		Set_zero				
		Syn_filt				
		Preemphasis				
	agc	energy_old				
		energy_new	energy_old			
		Inv_sqrt				
Post_Process						

4.5 Variables, constants and tables

The data types of variables and tables used in the fixed point implementation are signed integers in 2's complement representation, defined by:

Word16 16 bit variable
Word32 32 bit variable

Furthermore some **enum** types are used, all possible to represent with one byte, and a boolean **Flag**.

4.5.1 Description of constants used in the C-code

This section contains a listing of all global constants defined in `cnst.h`.

Table 5: Global constants

Constant	Value	Description
L_TOTAL	320	total size of speech buffer.
L_WINDOW	240	window size in LP analysis
L_FRAME	160	frame size
L_FRAME_BY2	80	frame size divided by 2
L_SUBFR	40	subframe size
L_CODE	40	codevector length
NB_TRACK	5	number of tracks
STEP	5	codebook step size
NB_TRACK_MR102	4	number of tracks mode mr102
STEP_MR102	4	codebook step size mode mr102
M	10	order of LP filter
MP1	(M+1)	order of LP filter + 1
LSF_GAP	205	minimum distance between LSF after quantization; 50 Hz = 205
LSP_PRED_FAC_MR122	21299	MR122 LSP prediction factor (0.65 Q15)
AZ_SIZE	44	size of array of LP filters in 4 subframes (4*M+4)
PIT_MIN_MR122	18	minimum pitch lag (MR122 mode)
PIT_MIN	20	minimum pitch lag (all other modes)
PIT_MAX	143	maximum pitch lag
L_INTERPOL	(10+1)	length of filter for interpolation
L_INTER_SRCH	4	length of filter for CL LTP search interpolation
MU	26214	factor for tilt compensation filter 0.8
AGC_FAC	29491	factor for automatic gain control 0.9
L_NEXT	40	overhead in LP analysis
SHARPMAX	13017	maximum value of pitch sharpening
SHARPMIN	0	minimum value of pitch sharpening
MAX_PRM_SIZE	57	max. num. of params
MAX_SERIAL_SIZE	244	max. num. of serial bits
GP_CLIP	15565	pitch gain clipping = 0.95
N_FRAME	7	old pitch gains in average calculation
EHF_MASK	8	16 bit representation of all samples in the encoder homing frame (left justification)

4.5.2 Description of fixed tables used in the C-code

This section contains a listing of all fixed tables sorted by source file name and table name. All table data is declared as **Word16**.

Table 6: Fixed tables

File	Table name	Length	Description
c2_9pf.c	trackTable	4*5	track table for algebraic code book search (MR475, MR515)
cod_amr.c	gamma1	10	spectral expansion factors
cod_amr.c	gamma1_12k2	10	spectral expansion factors
cod_amr.c	gamma2	10	spectral expansion factors
dtx_dec.c	lsf_hist_mean_scale	10	initialization values for DTX lsf parameters
dtx_dec.c	dtx_log_en_adjust	9	level adjustments for ech mode
ec_gains.c	cdown	7	attenuation factors for codebook gain
ec_gains.c	pdown	7	attenuation factors for adaptive codebook gain
gc_pred.c	pred	4	algebraic code book gain MA predictor coefficients
gc_pred.c	pred_MR122	4	algebraic code book gain MA predictor coefficients (MR122)
pitch_fr.c	mode_dep_parm	72	parameters defining the adaptive codebook search per mode
post_pro.c	a	3	HP filter coefficients (denominator) in Post_Process
post_pro.c	b	3	HP filter coefficients (numerator) in Post_Process
pre_proc.c	a	3	HP filter coefficients (denominator) in Pre_Process
pre_proc.c	b	3	HP filter coefficients (numerator) in Pre_Process
pred_lt.c	inter_6	61	interpolation filter coefficients
pstfilt.c	gamma3_MR122	10	spectral expansion factors
pstfilt.c	gamma3	10	spectral expansion factors
pstfilt.c	gamma4_MR122	10	spectral expansion factors
pstfilt.c	gamma4	10	spectral expansion factors
bitno.tab	prmno	9	number of bits for each mode
bitno.tab	prmnofs	8	number of parameters for LPC and first subframe for each mode (used for decoder homing procedure)
bitno.tab	bitno	9	pointers to the bitno_MR... tables
bitno.tab	bitno_MR475	17	number of bits per parameter to transmit (MR475)
bitno.tab	bitno_MR515	19	number of bits per parameter to transmit (MR515)
bitno.tab	bitno_MR59	19	number of bits per parameter to transmit (MR59)
bitno.tab	bitno_MR67	19	number of bits per parameter to transmit (MR67)
bitno.tab	bitno_MR74	19	number of bits per parameter to transmit (MR74)
bitno.tab	bitno_MR795	23	number of bits per parameter to transmit (MR795)
bitno.tab	bitno_MR102	39	number of bits per parameter to transmit (MR102)
bitno.tab	bitno_MR122	57	number of bits per parameter to transmit (MR122)
bitno.tab	bitno_MRDTX	5	number of bits per parameter to transmit (MRDTX)
c2_11pf.tab	startPos1	2	track start search position for first pulse
c2_11pf.tab	startPos2	4	track start search position for second pulse
c2_9pf.tab	startPos	16	track start search position
corrwgth.tab	corrweight	251	weighting of the correlation function in open loop LTP search (MR102)
d_homing.tab	dhf	8	pointers to the dhf_MR... tables
d_homing.tab	dhf_MR475	17	parameter values for the decoder homing frame (MR475)
d_homing.tab	dhf_MR515	19	parameter values for the decoder homing frame (MR515)
d_homing.tab	dhf_MR59	19	parameter values for the decoder homing frame (MR59)
d_homing.tab	dhf_MR67	19	parameter values for the decoder homing frame (MR67)
d_homing.tab	dhf_MR74	19	parameter values for the decoder homing frame (MR74)
d_homing.tab	dhf_MR795	23	parameter values for the decoder homing frame (MR795)
d_homing.tab	dhf_MR102	39	parameter values for the decoder homing frame (MR102)
d_homing.tab	dhf_MR122	57	parameter values for the decoder homing frame (MR122)
gains.tab	qua_gain_pitch	16	adaptive codebook gain quantization table (MR122, MR795)
gains.tab	qua_gain_code	96	fixed codebook gain quantization table (MR122, MR795)
gray.tab	gray	8	gray coding table
gray.tab	dgray	8	gray decoding table
grid.tab	grid	61	grid points at which Chebyshev polynomials are evaluated
inter_36.tab	inter_6	25	interpolation filter coefficients
inv_sqrt.tab	table	49	table used in inverse square root computation
lag_wind.tab	lag_h	10	high part of the lag window table
lag_wind.tab	lag_l	10	low part of the lag window table

(continued)

Table 6 (concluded): Fixed tables

File	Table name	Length	Description
------	------------	--------	-------------

log2.tab	table	33	table used inbase 2 logarithm computation
lsp.tab	lsp_init_data	10	initialization table for lsp history in DTX
lsp_lsf.tab	table	65	table to compute cos(x) in Lsf_lsf()
lsp_lsf.tab	slope	64	table to compute acos(x) in Lsp_lsf()
ph_disp.tab	ph_imp_low_MR795	40	phase dispersion impulse response (MR795)
ph_disp.tab	ph_imp_mid_MR795	40	phase dispersion impulse response (MR795)
ph_disp.tab	ph_imp_low	40	phase dispersion impulse response (MR475 - MR67)
ph_disp.tab	ph_imp_mid	40	phase dispersion impulse response (MR475 - MR67)
pow2.tab	table	33	table used in 2 to the power computation
q_plsf_3.tab	past_rq_init	80	initialization table for the MA predictor in DTX
q_plsf_3.tab	mean_lsf	10	LSF means (not in MR122)
q_plsf_3.tab	pred_fac	10	LSF prediction factors (not in MR122)
q_plsf_3.tab	dico1_lsf	3*256	1 st LSF quantizer (not in MR122 and MR795)
q_plsf_3.tab	dico2_lsf	3*512	2 nd LSF quantizer (not in MR122)
q_plsf_3.tab	dico3_lsf	4*512	3 rd LSF quantizer (not in MR122, MR515 and MR475)
q_plsf_3.tab	mr515_3_lsf	4*128	3 rd LSF quantizer (MR515 and MR475)
q_plsf_3.tab	mr795_1_lsf	3*512	1 st LSF quantizer (MR795)
q_plsf_5.tab	mean_lsf	10	LSF means (MR122)
q_plsf_5.tab	dico1_lsf	4*128	1 st LSF quantizer (MR122)
q_plsf_5.tab	dico2_lsf	4*256	2 nd LSF quantizer (MR122)
q_plsf_5.tab	dico3_lsf	4*256	3 rd LSF quantizer (MR122)
q_plsf_5.tab	dico4_lsf	4*256	4 th LSF quantizer (MR122)
q_plsf_5.tab	dico5_lsf	4*64	5 th LSF quantizer (MR122)
qgain475.tab	table_gain_MR475	4*256	gain quantization table (MR475)
qua_gain.tab	table_gain_highrates	128*4	gain quantization table (MR67, MR74 and MR102)
qua_gain.tab	table_gain_lowrates	64*4	gain quantization table (MR515 and MR59)
R_fft.c	phs_tbl	128	sine/cosine phase table
R_fft.c	ii_table	8	indexing table
sqrt_l	table	49	table to compute sqrt(x)
Vad1.c	ch_tbl	2*16	channel energy combination table
Vad1.c	ch_tbl_sh	16	channel energy scaling table
Vad1.c	vm_tbl	90	voice metric table
Vad1.c	hangover_table	20	used to determine hangover as a function of SNR
Vad1.c	burstcount_table	20	used to determine burst count threshold as a function of SNR
Vad1.c	vm_thresh_table	20	used to determine the voice metric threshold as a function of SNR
Vad1.c	energy state tables	2*6	constants as a function of scaling state
window.tab	window_200_40	240	LP analysis window (not in MR122)
window.tab	window_160_80	240	1 st LP analysis window (MR122)
window.tab	window_232_8	240	2 nd LP analysis window (MR122)

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
Pre_ProcessState	complexityCounter	int	Used for wMOPS counting
	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
cod_amrState	x1	Word16	filter state
	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
	vadSt1	vadState1	see below in this table
	VadSt2	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
	VadState1	bckr_est	Word16[9]
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
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_engr_long_db	Word16[16]	long term channel energy in dB
	Lframe_cnt	Word32	10 ms frame counter
	Lch_engr	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
	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	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	weighthing 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	Decoder_amrState	Decoder_amrState	see below in this table
	post_state PostHP_state	Post_FilterState Post_ProcessState	see below in this table 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_avgState	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	
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	Word16[5]	pitch gain history
	prevState	Word16	previously used impulse response
	prevCbGain	Word16	previous codebook gain
	lockFull	Word16	force maximum phase dispersion
Post_FilterState	onset	Word16	onset counter
	res2	Word16[40]	LP residual
	mem_syn_pst	Word16[10]	synthesis filter memory
	synth_buf	Word16[170]	synthesis filter work area
	agc_state	agcState	see below in this table
preemphasisState	preemph_state	preemphasisState	see below in this table
	past_gain	Word16	past agc gain
Post_ProcessState	mem_pre	Word16	filter state
Post_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

5 Homing procedure

The principles of the homing procedures are described in TS 06.090 [2]. This specification only includes a detailed description of the 8 decoder homing frames. For each AMR codec mode, the corresponding decoder homing frame has a fixed set of speech parameters shown in table 9a-9h. The bit allocation within these parameters is identical to the corresponding bit allocation of the source encoder output parameters given in TS 06.090 [2].

In the following tables, the following naming convention is used for the individual parameters. Letters in *italics>* indicate numbers.

- LPC_{*n*} index of *n*th LSF submatrix
- LTP-LAG_{*m*} adaptive codebook index for subframe *m*
- LTP-GAIN_{*m*} adaptive codebook gain index in subframe *m*
- FCB-GAIN_{*m*} fixed codebook gain index in subframe *m*
- GAIN_VQ_{*m*} codebook gain VQ index in subframe *m* (subframe *m* and *m+1* for MR475)
- POS_{*m_n*} position index of *n*th pulse in subframe *m*
- POS_{*m_n_k*} position index of *n*th and *k*th pulse in subframe *m*
- POS_{*m_n_k_l_j*} position index of *n*th, *k*th, *l*th, and *j*th pulse in subframe *m*
- SIGN_{*m_n_k*} sign information for *n*th and *k*th pulse in subframe *m*
- SIGN_{*m_n_k_l_j*} sign information for *n*th, *k*th, *l*th, and *j*th pulse in subframe *m*
- SIGN_{*m_n_k_POS_m_n*} sign information for *n*th and *k*th pulse and position index for *n*th pulse in subframe *m*

Table 9a: Parameter values for the decoder homing frame (MR475)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x009D
LPC 3	0x001C
LTP-LAG 1	0x0066
POS 1_1_2	0x0000
SIGN 1_1_2	0x0003
GAIN-VQ 1	0x0028
LTP-LAG 2	0x000F
POS 2_1_2	0x0038
SIGN 2_1_2	0x0001
LTP-LAG 3	0x000F
POS 3_1_2	0x0031
SIGN 3_1_2	0x0002
GAIN-VQ 3	0x0008
LTP-LAG 4	0x000F
POS 4_1_2	0x0026
SIGN 4_1_2	0x0003

Table 9b: Parameter values for the decoder homing frame (MR515)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x009D
LPC 3	0x001C
LTP-LAG 1	0x0066
POS 1_1_2	0x0000
SIGN_1_1_2	0x0003
GAIN-VQ 1	0x0037
LTP-LAG 2	0x000F
POS 2_1_2	0x0000
SIGN_2_1_2	0x0003
GAIN-VQ 2	0x0005
LTP-LAG 3	0x000F
POS 3_1_2	0x0037
SIGN_3_1_2	0x0003
GAIN-VQ 3	0x0037
LTP-LAG 4	0x000F
POS 4_1_2	0x0023
SIGN_4_1_2	0x0003
GAIN-VQ 4	0x001F

Table 9c: Parameter values for the decoder homing frame (MR59)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x00E3
LPC 3	0x002F
LTP-LAG 1	0x00BD
POS 1_1_2	0x0000
SIGN_1_1_2	0x0003
GAIN-VQ 1	0x0037
LTP-LAG 2	0x000F
POS 2_1_2	0x0001
SIGN_2_1_2	0x0003
GAIN-VQ 2	0x000F
LTP-LAG 3	0x0060
POS 3_1_2	0x00F9
SIGN_3_1_2	0x0003
GAIN-VQ 3	0x0037
LTP-LAG 4	0x000F
POS 4_1_2	0x0000
SIGN_4_1_2	0x0003
GAIN-VQ 4	0x0037

Table 9d: Parameter values for the decoder homing frame (MR67)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x00E3
LPC 3	0x002F
LTP-LAG 1	0x00BD
POS 1_1_2_3	0x0002
SIGN_1_1_2_3	0x0007
GAIN-VQ 1	0x0000
LTP-LAG 2	0x000F
POS 2_1_2_3	0x0098
SIGN_2_1_2_3	0x0007
GAIN-VQ 2	0x0061
LTP-LAG 3	0x0060
POS 3_1_2_3	0x05C5
SIGN_3_1_2_3	0x0007
GAIN-VQ 3	0x0000
LTP-LAG 4	0x000F
POS 4_1_2_3	0x0318
SIGN_4_1_2_3	0x0007
GAIN-VQ 4	0x0000

Table 9e: Parameter values for the decoder homing frame (MR74)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x00E3
LPC 3	0x002F
LTP-LAG 1	0x00BD
POS 1_1_2_3_4	0x0006
SIGN_1_1_2_3_4	0x000F
GAIN-VQ 1	0x0000
LTP-LAG 2	0x001B
POS 2_1_2_3_4	0x0208
SIGN_2_1_2_3_4	0x000F
GAIN-VQ 2	0x0062
LTP-LAG 3	0x0060
POS 3_1_2_3_4	0x1BA6
SIGN_3_1_2_3_4	0x000F
GAIN-VQ 3	0x0000
LTP-LAG 4	0x001B
POS 4_1_2_3_4	0x0006
SIGN_4_1_2_3_4	0x000F
GAIN-VQ 4	0x0000

Table 9f: Parameter values for the decoder homing frame (MR795)

Parameter	Value (LSB=b0)
LPC 1	0x00C2
LPC 2	0x00E3
LPC 3	0x002F
LTP-LAG 1	0x00BD
POS 1_1_2_3_4	0x0006
SIGN_1_1_2_3_4	0x000F
LTP-GAIN 1	0x000A
FCB-GAIN 1	0x0000
LTP-LAG 2	0x0039
POS 2_1_2_3_4	0x1C08
SIGN_2_1_2_3_4	0x0007
LTP-GAIN 2	0x000A
FCB-GAIN 2	0x000B
LTP-LAG 3	0x0063
POS 3_1_2_3_4	0x11A6
SIGN_3_1_2_3_4	0x000F
LTP-GAIN 3	0x0001
FCB-GAIN 3	0x0000
LTP-LAG 4	0x0039
POS 4_1_2_3_4	0x09A0
SIGN_4_1_2_3_4	0x000F
LTP-GAIN 4	0x0002
FCB-GAIN 4	0x0001

Table 9g: Parameter values for the decoder homing frame (MR102)

Parameter	Value (LSB=b0)
LPC 1	0x00F8
LPC 2	0x00E3
LPC 3	0x002F
LTP-LAG 1	0x0045
SIGN_1_1_5	0x0000
SIGN_1_2_6	0x0000
SIGN_1_3_7	0x0000
SIGN_1_4_8	0x0000
POS_1_1_2_5	0x0000
POS_1_3_6_7	0x0000
POS_1_4_8	0x0000
GAIN-VQ_1	0x0000
LTP-LAG 2	0x001B
SIGN_2_1_5	0x0000
SIGN_2_2_6	0x0001
SIGN_2_3_7	0x0000
SIGN_2_4_8	0x0001
POS_2_1_2_5	0x0326
POS_2_3_6_7	0x00CE
POS_2_4_8	0x007E
GAIN-VQ_2	0x0051
LTP-LAG 3	0x0062
SIGN_3_1_5	0x0000
SIGN_3_2_6	0x0000
SIGN_3_3_7	0x0000
SIGN_3_4_8	0x0000
POS_3_1_2_5	0x015A
POS_3_3_6_7	0x0359
POS_3_4_8	0x0076
GAIN-VQ_3	0x0000
LTP-LAG 4	0x001B
SIGN_4_1_5	0x0000
SIGN_4_2_6	0x0000
SIGN_4_3_7	0x0000
SIGN_4_4_8	0x0000
POS_4_1_2_5	0x017C
POS_4_3_6_7	0x0215
POS_4_4_8	0x0038
GAIN-VQ_4	0x0030

Table 9h: Parameter values for the decoder homing frame (MR122)

Parameter	Value (LSB=b0)
LPC1	0x0004
LPC2	0x002A
LPC3	0x00DB
LPC4	0x0096
LPC5	0x002A
LTP-LAG 1	0x0156
LTP-GAIN 1	0x000B
SIGN_1_1_6_POS_1_1	0x0000
SIGN_1_2_7_POS_1_2	0x0000
SIGN_1_3_8_POS_1_3	0x0000
SIGN_1_4_9_POS_1_4	0x0000
SIGN_1_5_10_POS_1_5	0x0000
POS 1_6	0x0000
POS 1_7	0x0000
POS 1_8	0x0000
POS 1_9	0x0000
POS 1_10	0x0000
FCB-GAIN 1	0x0000
LTP-LAG 2	0x0036
LTP-GAIN 2	0x000B
SIGN_2_1_6_POS_2_1	0x0000
SIGN_2_2_7_POS_2_2	0x000F
SIGN_2_3_8_POS_2_3	0x000E
SIGN_2_4_9_POS_2_4	0x000C
SIGN_2_5_10_POS_2_5	0x000D
POS 2_6	0x0000
POS 2_7	0x0001
POS 2_8	0x0005
POS 2_9	0x0007
POS 2_10	0x0001
FCB-GAIN 2	0x0008
LTP-LAG 3	0x0024
LTP-GAIN 3	0x0000
SIGN_3_1_6_POS_3_1	0x0001
SIGN_3_2_7_POS_3_2	0x0000
SIGN_3_3_8_POS_3_3	0x0005
SIGN_3_4_9_POS_3_4	0x0006
SIGN_3_5_10_POS_3_5	0x0001
POS 3_6	0x0002
POS 3_7	0x0004
POS 3_8	0x0007
POS 3_9	0x0004
POS 3_10	0x0002
FCB-GAIN 3	0x0003
LTP-LAG 4	0x0036
LTP-GAIN 4	0x000B
SIGN_4_1_6_POS_4_1	0x0000
SIGN_4_2_7_POS_4_2	0x0002
SIGN_4_3_8_POS_4_3	0x0004
SIGN_4_4_9_POS_4_4	0x0000
SIGN_4_5_10_POS_4_5	0x0003
POS 4_6	0x0006
POS 4_7	0x0001
POS 4_8	0x0007
POS 4_9	0x0006
POS 4_10	0x0005
FCB-GAIN 4	0x0000

6 File formats

This section describes the file formats used by the encoder and decoder programs. The test sequences defined in [2] also use the file formats described here.

6.1 Speech file (encoder input / decoder output)

Speech files read by the encoder and written by the decoder consist of 16-bit words where each word contains a 13-bit, left aligned speech sample. The byte order depends on the host architecture (e.g. MSByte first on SUN workstations, LSByte first on PCs etc.). Both the encoder and the decoder program process complete frames (of 160 samples) only. This means that the encoder will only process n frames if the length of the input file is $n*160 + k$ words, while the files produced by the decoder will always have a length of $n*160$ words,

6.2 Mode control file (encoder input)

The encoder program can optionally read in a mode control file which specifies the encoding mode for each frame of speech processed. The file is a text file containing one line per speech frame. Each line contains one of the mode names from the list {MR475, MR515, MR59, MR67, MR74, MR795, MR102, MR122}.

6.3 Parameter bitstream file (encoder output / decoder input)

The files produced by the speech encoder/expected by the speech decoder contain an arbitrary number of frames in the following format:

FRAME_TYPE	B1	B2	...	B244	MODE_INFO	<i>unused1</i>	...	<i>unused4</i>
------------	----	----	-----	------	-----------	----------------	-----	----------------

Each box corresponds to one Word16 value in the bitstream file, for a total of 250 words or 500 bytes per frame. The fields have the following meaning:

FRAME_TYPE	transmit frame type, which is one of
	TX_SPEECH (0x0000)
	TX_SID_FIRST (0x0001)
	TX_SID_UPDATE (0x0002)
	TX_NO_DATA (0x0003)
B0...B244	speech encoder parameter bits (i.e. the bitstream itself). Each B _x either has the value 0x0000 or 0x0001. Only mode MR122 really uses all 244 bits; for the other modes, only the first n bits are used ($35 \leq n \leq 204$). The remaining bits are unused (written as 0x0000)
MODE_INFO	encoding mode information, which is one of
	MR475 (0x0000)
	MR515 (0x0001)
	MR59 (0x0002)
	MR67 (0x0003)
	MR74 (0x0004)
	MR795 (0x0005)
	MR102 (0x0006)
	MR122 (0x0007)
<i>unused1...4</i>	unused, written as 0x0000

As indicated in section 6.1 above, the byte order depends on the host architecture.

History

Document history		
V. 0.1.0	March 1999	First Draft
V. 0.5.0	September 1999	Technical content copied from GSM 06.73 v 7.1.0
V. 1.0.0	October 1999	Presented for information to TSG SA#5 Plenary