RECOMMENDATION ITU-R M.1309*

DIGITALLY CODED SPEECH IN THE LAND MOBILE SERVICE

(Question ITU-R 101/8)

(1997)

Summary

Many new digital systems have been developed which are rapidly overtaking analogue systems for land mobile and cordless telephone voice communications. This Recommendation provides a brief description of speech coding technologies utilized in current and emerging systems.

The ITU Radiocommunication Assembly,

considering

a) that there is rapid development in methods for the digitization of speech;

b) that different methods of speech coding using differing bit rates and providing various qualities of speech are available;

c) that the perceived quality of speech when using digitized speech transmission should be equal to or better than analogue in a comparable application;

d) that the requirements on speech codec (coder/decoder) performance may differ between various applications;

e) that there may be advantages in adopting the land mobile service standards that are compatible with ITU-T Recommendations relevant to the fixed networks;

f) that different methods of optimizing coding of speech and channel coding or modulation altogether are available;

g) that the ability of codecs to adapt their rates dynamically to varying transmission conditions could lead to significant spectrum efficiency gains;

h) that the design of the codec has an effect on the power requirements;

j) that there may be a need to compare several proposed codecs for any given application,

recommends

1 that the requirements on speech codec should be carefully matched to the application, taking into account the desired speech quality, the spectrum available for the system, the traffic capacity and other factors, such as the power consumption of the equipment;

2 that codecs have as low a bit rate as possible to give the required radio system performance without the need for excessive forward error correction (FEC). The combined codec and FEC rate must be considered when assessing spectrum efficiency;

3 that when appropriate, costs and development times can be reduced by using previously adopted standards, further work should be encouraged to develop such standards;

4 that where possible, codecs should make use of techniques, such as, varying bit rates to dynamically adapt to the transmission conditions;

5 that a methodology for assessing and quantifying the performance of codecs should be developed to facilitate fair and equitable comparisons;

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* This Recommendation should be brought to the attention of the Telecommunication Standardization Sector (ITU-T).

5.1 the methodology should facilitate the selection of a codec taking into account the appropriate weightings for all system parameters associated with codecs. Some of the parameters that may be considered are:

– speech performance;

– codec bit rate;

– codec complexity and cost;

– processing power required to maintain real-time function;

– the ability of the codec rate/FEC to adapt to varying radio environments, and the effect this has on spectrum efficiency and battery duty cycle/power consumption;

– the compatibility of the codec with other radio systems (e.g. fixed);

– the ease with which encryption can be added;

– robustness;

– performance in the presence of other signals and of background noise;

– delay of the coder;

5.2 that consideration should be given to the above parameters to be used to construct a matrix which compares codec features pairwise as outlined in the International Mobile Telecommunications (IMT-2000) selection methodology (See Recommendation ITU-R M.1225). The correct implementation of the methodology would allow a framework to be designed to ensure the best system was selected for any given application;

6 that Annex 1 contains examples of those codecs used in the land mobile service which were presented to the ITU-R; they are given here for information.

ANNEX 1

Speech coding systems used in the land mobile service

	Cellular (TDMA)		
System name	GSM/DCS/PCS (Full rate)	GSM/DCS/PCS (Half rate)	GSM EFR US PCS 1900 EFR
Origin	ETSI	ETSI	USA/ETSI
Frequency (GHz)	0.9/1.8/1.9	0.9/1.8/1.9	0.9/1.9
Modulation	GMSK	GMSK	GMSK
Maximum gross trns rate (kbit/s)(1)	33.9	16.95	29.6
Codec Type	RPE-LTP	VSELP	ACELP
Codec rate (kbit/s)	13	5.6	12.2
FEC (kbit/s)(2)	9.8	5.8	10.6
Codec Algorithmic delay (ms)(3)	40	40	40
Radio System delay (ms)(4)	50	50	50
Estimated Quality (MOS)(5), (6)	3.6-3.8	3.5-3.7	4.1
Estimated Processing(7)	2.5 Mips	17.5 Mips	15.4 WMops
Reference Document	ETSI/ETS 300580 ANSI J-STD-007	ETSI/ETS 300581	ANSI J-STD-007A ETSI/ETS 300723

	Cellular (TDMA)			
System name	D-AMPS (Full rate)	D-AMPS (Enhanced)	PDC (Full rate)	PDC (Half rate)
Origin	USA	USA	Japan	Japan
Frequency (GHz)	0.8	0.8/1.9	0.8-0.9/1.5	0.8-0.9/1.5
Modulation	π/4 DQPSK	π/4 DQPSK	π/4 DQPSK	π/4 DQPSK
Maximum gross trns rate (kbit/s)(1)	16.2	16.2	14	7.0
Codec Type	VSELP	ACELP	VSELP	PSI-CELP
Codec rate (kbit/s)	8.0	7.4	6.7	3.45
FEC (kbit/s)(2)	5.0	5.6	4.5	2.15
Codec Algorithmic delay (ms)(3)	28	25	47	85
Radio System delay (ms)(4)	48	45	75	130
Estimated Quality (MOS)(5), (6)	3.7	4.1	3.40	3.34
Estimated Processing(7)	22 Wmops	14 Wmops	7.8 Mops	18.7 Mops
Reference Document	TIA/EIA IS-85	TIA/EIA IS-641	RCR STD-27	RCR STD-27

	Cellular (CDMA)		
System name	Composite CDMA/TDMA	CDMA (IS-96)	CDMA (IS-127)
Origin	USA	USA	USA
Frequency (GHz)	1.9	0.8-0.9/1.85-1.99	0.8-0.9/1.85-1.99
Modulation	OQPSK	QPSK	QPSK
Maximum gross trns rate (kbit/s)(1)	10.4	28.8	28.8
Codec Type	CELP like	CELP	RCELP
Codec rate (kbit/s)	7.2	8/4/2/0.8	8/4/0.8
FEC (kbit/s)(2)	3.2	19.2	19.2
Codec Algorithmic delay (ms)(3)	26	27.5	30
Radio System delay (ms)(4)	60	37.5	40
Estimated Quality (MOS)(5), (6)	> 4.0	3.3	4.1
Estimated Processing(7)	11 Mips	22 Mops	20 Mops
Reference Document	TIA/EIA IS-661	TIA/EIA IS-96	TIA/EIA IS-127

	Cordless		
System name	CT2	DECT	PHS
Origin	ETSI	ETSI	Japan
Frequency (GHz)	0.86	1.9	1.9
Modulation	GFSK	GFSK	π/4) DQPSK
Maximum gross trans rate (kbit/s)(1)	72	96	48
Codec Type	ADPCM(8)	ADPCM(8)	ADPCM(8)
Codec rate (kbit/s)	32	32	32
FEC (kbit/s)(2)	0	0	0
Codec Algorithmic delay (ms)(3)	(9)	(9)	(9)
Radio System delay (ms)(4)	0.375	0.375	0.25
Estimated Quality (MOS)(5), (6)	4.0	4.0	4.0
Estimated Processing (7)	10 Mips	10 Mips	1.0 Mops
Reference Document	Rec. ITU-T G.726	Rec. ITU-T G.726	Rec. ITU-T G.726

	Dispatch			
System name	Projet 25	TETRA	IDRA	DIMRS
Origin	USA	ETSI	Japan	Canada
Frequency (GHz)	0.15-0.9	0.4/0.9	0.8/1.5	0.8
Modulation	C4FM/CQPSK	π/4 DQPSK	M16QAM	M16QAM
Maximum gross trans rate (kbit/s)(1)	9.6	9.0	10.7	10.7/21.33
Codec Type	IMBE	ACELP	CELP/VSELP(10)	VSELP
Codec rate (kbit/s)	4.4	4.567	4.7/4.2(10)	4.2/8.0
FEC (kbit/s)(2)	2.8	2.633	2.766/3.177(10)	3.177/6.756
Codec Algorithmic delay (ms)(3)	80	75	81/87	75/45
Radio System delay (ms)(4)	120	90	96/102(10)	102/54.6
Estimated Quality (MOS)(5), (6)	3.4	3.3-3.5	3.2	3.20/3.98
Estimated Processing(7)	6.9 Mips	15 Mips	7.0/8.0 Mops(10)	8.0 Mops
Reference Document	TIA/EIA IS-102.BABA	ETSI/ETS 300395	RCR STD-32A	68P81129E15 Motorola

ACELP: algebraic code excited linear prediction
ADPCM: adaptive differential pulse code modulation
CDMA: code division multiple access
CELP: code excited linear prediction
CQPSK coherent quaternary phase shift keying
C4FM: constant-envelope 4-level frequency modulation (FM)

DQPSK: differential quadriphase pulse shift keying

GFSK: gaussian filtered frequency shift keying

GMSK: gaussian filtered minimum shift keying

IMBE: improved multiband excitation

Mips: millions of instructions per second (required by codec)

Mops: millions of operations per second (required by codec)

MOS mean opinion score

M16QAM: 16-states quadrature amplitude modulation

OQPSK: offset quaternary phase shift keying

PSI-CELP: peripheral subsystem interface – CELP

QPSK: quadraphase shift keying

RCELP: relaxed code excited linear prediction

RPE-LTP: regular pulse excitation linear predictive coding (LPC) using long-term prediction

TDMA: time division multiple access

VSELP: vector sum excited linear prediction

Wmops: weighted millions of operations per second (required by codec)

(1) Is the maximum bit rate available to be allocated to a single user.

(2) Includes only bits used for FEC excluding any other bits used for, such as, pilot or synchronization.

(3) Codec Algorithmic Delay = Buffering Delay + Voice Encode + Voice Decode Delay.

(4) Radio System Delay = Buffering Delay + Voice Encode Delay + FEC Encode Delay + Transmission Delay + FEC Decode Delay + Voice Decode Delay.

The Radio System Delay does not include delays due to repeaters and networking.

(5) Tested under no error condition.

(6) MOS scores for a given codec can and do vary from test to test, based on different languages, populations, test labs and different ensembles of codecs. MOS scores from different listening tests are not directly comparable. It is recommended for future study that a more standardized voice quality testing procedure be developed. (Refer to ITU-T Recommendations P.830 regarding subjective test and P.861 regarding objective test for guidance on test methods.)

(7) Processor dependant.

(8) ITU-T Recommendation G.726 at 32 kbit/s.

(9) Algorithmic delay is not applicable.

(10) These items are not standardized and two types of coding techniques are being used in Japan.