INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.722.2

Annex A
(01/2002)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

Digital terminal equipments – Coding of analogue signals by methods other than PCM

Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)

Annex A: Comfort noise aspects

ITU-T Recommendation G.722.2 – Annex A

ITU-T G-SERIES RECOMMENDATIONS

TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

INTERNATIONAL TELEPHONE CONNECTIONS AND CIRCUITS	G.100–G.199
GENERAL CHARACTERISTICS COMMON TO ALL ANALOGUE CARRIER-TRANSMISSION SYSTEMS	G.200–G.299
INDIVIDUAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON METALLIC LINES	G.300–G.399
GENERAL CHARACTERISTICS OF INTERNATIONAL CARRIER TELEPHONE SYSTEMS ON RADIO-RELAY OR SATELLITE LINKS AND INTERCONNECTION WITH METALLIC LINES	G.400–G.449
COORDINATION OF RADIOTELEPHONY AND LINE TELEPHONY	G.450–G.499
TESTING EQUIPMENTS	G.500–G.599
TRANSMISSION MEDIA CHARACTERISTICS	G.600–G.699
DIGITAL TERMINAL EQUIPMENTS	G.700–G.799
General	G.700–G.709
Coding of analogue signals by pulse code modulation	G.710–G.719
Coding of analogue signals by methods other than PCM	**G.720–G.729**
Principal characteristics of primary multiplex equipment	G.730–G.739
Principal characteristics of second order multiplex equipment	G.740–G.749
Principal characteristics of higher order multiplex equipment	G.750–G.759
Principal characteristics of transcoder and digital multiplication equipment	G.760–G.769
Operations, administration and maintenance features of transmission equipment	G.770–G.779
Principal characteristics of multiplexing equipment for the synchronous digital hierarchy	G.780–G.789
Other terminal equipment	G.790–G.799
DIGITAL NETWORKS	G.800–G.899
DIGITAL SECTIONS AND DIGITAL LINE SYSTEM	G.900–G.999
QUALITY OF SERVICE AND PERFORMANCE	G.1000–G.1999
TRANSMISSION MEDIA CHARACTERISTICS	G.6000–G.6999
General	G.6000–G.6099
Symmetric cable pairs	G.6100–G.6199
Land coaxial cable pairs	G.6200–G.6299
Submarine cables	G.6300–G.6499
Optical fibre cables	G.6500–G.6599
Characteristics of optical components and subsystems	G.6600–G.6999
DIGITAL TERMINAL EQUIPMENTS	G.7000–G.7999
General	G.7000–G.7099
Coding of analogue signals by pulse code modulation	G.7100–G.7199
Coding of analogue signals by methods other than PCM	G.7200–G.7299
Principal characteristics of primary multiplex equipment	G.7300–G.7399
Principal characteristics of second order multiplex equipment	G.7400–G.7499
Principal characteristics of higher order multiplex equipment	G.7500–G.7599
Principal characteristics of transcoder and digital multiplication equipment	G.7600–G.7699
Operations, administration and maintenance features of transmission equipment	G.7700–G.7799
Principal characteristics of multiplexing equipment for the synchronous digital hierarchy	G.7800–G.7899
Other terminal equipment	G.7900–G.7999
DIGITAL NETWORKS	G.8000–G.8999
General aspects	G.8000–G.8099
Design objectives for digital networks	G.8100–G.8199
Quality and availability targets	G.8200–G.8299
Network capabilities and functions	G.8300–G.8399
SDH network characteristics	G.8400–G.8499
Management of transport network	G.8500–G.8599
SDH radio and satellite systems integration	G.8600–G.8699
Optical transport networks	G.8700–G.8799

For further details, please refer to the list of ITU-T Recommendations.

ITU-T Recommendation G.722.2

Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)

Annex A

Comfort noise aspects

Summary

This annex details the operation of the background acoustic noise evaluation, noise parameter encoding/decoding and comfort noise generation for the AMR Wideband (AMR-WB) speech codec during Source Controlled Rate (SCR) operation.

The comfort noise operations described here were also adopted by 3GPP in 3GPP specification TS 26.192.

Source

Annex A to ITU-T Recommendation G.722.2 was prepared by ITU-T Study Group 16 (2001-2004) and approved under the WTSA Resolution 1 procedure on 13 January 2002.

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications. The ITU Telecommunication Standardization Sector (ITU-T) is a permanent organ of ITU. ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Assembly (WTSA), which meets every four years, establishes the topics for study by the ITU-T study groups which, in turn, produce Recommendations on these topics.

The approval of ITU-T Recommendations is covered by the procedure laid down in WTSA Resolution 1.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

NOTE

In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

INTELLECTUAL PROPERTY RIGHTS

ITU draws attention to the possibility that the practice or implementation of this Recommendation may involve the use of a claimed Intellectual Property Right. ITU takes no position concerning the evidence, validity or applicability of claimed Intellectual Property Rights, whether asserted by ITU members or others outside of the Recommendation development process.

As of the date of approval of this Recommendation, ITU had received notice of intellectual property, protected by patents, which may be required to implement this Recommendation. However, implementors are cautioned that this may not represent the latest information and are therefore strongly urged to consult the TSB patent database.

CONTENTS

		Page
A.1	Scope	1
A.2	Definitions, symbols and abbreviations	1
A.2.1	Definitions	1
A.2.2	Symbols	1
A.2.3	Abbreviations	2
A.3	General	2
A.4	Functions on the transmit (TX) side	2
A.4.1	ISF evaluation	3
A.4.2	Frame energy calculation	4
A.4.3	Analysis of the variation and stationarity of the background noise	5
A.4.4	Modification of the speech encoding algorithm during SID frame generation	5
A.4.5	SID-frame encoding	6
A.5	Functions on the receive (RX) side	6
A.5.1	Averaging and decoding of the LP and energy parameters	6
A.5.2	Comfort noise generation and updating	7
A.6	Computational details and bit allocation	7

ITU-T Recommendation G.722.2

Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)

Annex A

Comfort noise aspects

A.1 Scope

This annex details the operation of the background acoustic noise evaluation, noise parameter encoding/decoding and comfort noise generation for the AMR Wideband (AMR-WB) speech codec during Source Controlled Rate (SCR) operation.

Implementation of this annex is necessary for interoperability with 3GPP systems, but its use is not limited to mobile applications.

The user should note however that the C-code implementation of this annex is available as part of the C-code in Annex C/G.722.2. In case of discrepancy between the requirements described in this annex and the fixed point computational description of these requirements contained in Annex C/G.722.2, the description in Annex C/G.722.2 shall prevail.

A.2 Definitions, symbols and abbreviations

A.2.1 Definitions

This annex defines the following terms:

A.2.1.1 frame: Time interval of 20 ms corresponding to the time segmentation of the adaptive multi-rate wideband speech transcoder, also used as a short term for "traffic frame".

A.2.1.2 SID frames: Special Comfort Noise frames. It may convey information on the acoustic background noise or inform the decoder that it should start generating background noise.

A.2.1.3 speech frame: Traffic frame that cannot be classified as a SID frame.

A.2.1.4 VAD flag: Voice Activity Detection flag.

A.2.1.5 TX_TYPE: Classification of the transmitted traffic frame (defined in Annex B/G.722.2).

A.2.1.6 RX_TYPE: Classification of the received traffic frame (defined in Annex B/G.722.2).

Other definitions of terms used in this annex can be found in the main body of ITU-T Rec. G.722.2 and in Annex B/G.722.2. The overall operation of SCR is described in Annex B/G.722.2.

A.2.2 Symbols

This annex uses the following symbols. Boldface symbols are used for vector variables:

$\mathbf{f}^T = [f_1 f_2 \ldots f_{16}]$	Unquantized ISF vector
$\hat{\mathbf{f}}^T = \left[\hat{f}_1 \hat{f}_2 \ldots \hat{f}_{16}\right]$	Quantized ISF vector
$\mathbf{f}^{(m)}$	Unquantized ISF vector of frame *m*
$\hat{\mathbf{f}}^{(m)}$	Quantized ISF vector of frame *m*
$\mathbf{f}^{mean}$	Averaged ISF parameter vector
$en_{\log}$	Logarithmic frame energy

$en_{\log}^{mean}$	Averaged logarithmic frame energy
$\mathbf{e}$	ISF parameter prediction residual
$\hat{\mathbf{e}}$	Quantized ISF parameter prediction residual
$\sum_{n=a}^{b} x(n)$	$= x(a) + x(a+1) + ... + x(b-1) + x(b)$

A.2.3 Abbreviations

This annex uses the following abbreviations:

AMR	Adaptive Multi-Rate
AMR-WB	Adaptive Multi-Rate Wideband
CN	Comfort Noise
ISF	Immittance Spectral Frequency
ISP	Immittance Spectral Pair
LP	Linear Prediction
RSS	Radio Subsystem
RX	Receive
SCR	Source Controlled Rate (operation) (aka source discontinuous transmission)
SID	Silence Insertion Descriptor
TX	Transmit
UE	User Equipment
VAD	Voice Activity Detector

A.3 General

A basic problem when using SCR is that the background acoustic noise, which is transmitted together with the speech, would disappear when the transmission is cut, resulting in discontinuities of the background noise. Since the SCR switching can take place rapidly, it has been found that this effect can be very annoying for the listener – especially in a car environment with high background noise levels. In bad cases, the speech may be hardly intelligible.

This annex specifies the way to overcome this problem by generating on the receive (RX) side synthetic noise similar to the transmit (TX) side background noise. The comfort noise parameters are estimated on the TX side and transmitted to the RX side at a regular rate when speech is not present. This allows the comfort noise to adapt to the changes of the noise on the TX side.

A.4 Functions on the transmit (TX) side

The comfort noise evaluation algorithm uses the following parameters of the AMR-WB speech encoder, defined in the main body of ITU-T Rec. G.722.2:

– the unquantized Linear Prediction (LP) parameters, using the Immittance Spectral Pair (ISP) representation, where the unquantized Immittance Spectral Frequency (ISF) vector is given by $\mathbf{f}^{\mathrm{T}} = [f_1 f_2 \ldots f_{16}]$;

The algorithm computes the following parameters to assist in comfort noise generation:

– the weighted averaged ISF parameter vector $\mathbf{f}^{mean}$ (weighted average of the ISF parameters of the eight most recent frames);

– the averaged logarithmic frame energy $en_{\log}^{mean}$ (average of the logarithmic energy of the eight most recent frames).

These parameters give information on the level ($en_{\log}^{mean}$) and the spectrum ($\mathbf{f}^{mean}$) of the background noise.

The evaluated comfort noise parameters ($\mathbf{f}^{mean}$ and $en_{\log}^{mean}$) are encoded into a special frame, called a Silence Insertion Descriptor (SID) frame, for transmission to the RX side.

A hangover logic is used to enhance the quality of the silence descriptor frames. A hangover of seven frames is added to the VAD flag so that the coder waits with the switch from active to inactive mode for a period of seven frames; during that time the decoder can compute a silence descriptor frame from the quantized ISFs and the logarithmic frame energy of the decoded speech signal. Therefore, no comfort noise description is transmitted in the first SID frame after active speech. If the background noise contains transients which will cause the coder to switch to active mode and then back to inactive mode in a very short time period, no hangover is used. Instead the previously used comfort noise frames are used for comfort noise generation.

The first SID frame also serves to initiate the comfort noise generation on the receive side, as a first SID frame is always sent at the end of a speech burst, i.e. before the transmission is terminated.

The scheduling of SID or speech frames on the network path is described in Annex B/G.722.2.

A.4.1 ISF evaluation

The comfort noise parameters to be encoded into a SID frame are calculated over $N = 8$ consecutive frames marked with VAD = 0, as follows:

Prior to averaging the ISF parameters over the CN averaging period, a median replacement is performed on the set of ISF parameters to be averaged, to remove the parameters which are not characteristic of the background noise on the transmit side. First, the spectral distances from each of the ISF parameter vectors $\mathbf{f}(i)$ to the other ISF parameter vectors $\mathbf{f}(j)$, i=0,...,7, j=0,...,7, $i \neq j$, within the CN averaging period are approximated according to the equation:

$$\Delta R_{ij} = \sum_{k=1}^{16} \left(f_i(k) - f_j(k) \right)^2 \qquad \text{(A-1)}$$

where $f_i(k)$ is the kth ISF parameter of the ISF parameter vector $\mathbf{f}(i)$ at frame i.

To find the spectral distance ΔS_i of the ISF parameter vector $\mathbf{f}(i)$ to the ISF parameter vectors $\mathbf{f}(j)$ of all the other frames j=0,...,7, $j \neq i$, within the CN averaging period, the sum of the spectral distances ΔR_{ij} is computed as follows:

$$\Delta S_i = \sum_{j=0, j \neq i}^{7} \Delta R_{ij} \qquad \text{(A-2)}$$

for all i=0,...,7, $i \neq j$.

The ISF parameter vector $\mathbf{f}(i)$ with the smallest spectral distance ΔS_i of all the ISF parameter vectors within the CN averaging period is considered as the median ISF parameter vector $\mathbf{f}_{med}$ of the averaging period, and its spectral distance is denoted as ΔS_{med}. The median ISF parameter vector is considered to contain the best representation of the short-term spectral detail of the background noise of all the ISF parameter vectors within the averaging period. If there are ISF parameter vectors $\mathbf{f}(j)$ within the CN averaging period with:

$$\frac{\Delta S_j}{\Delta S_{med}} > TH_{med} \tag{A-3}$$

where $TH_{med} = 2.25$ is the median replacement threshold, then at most two of these ISF parameter vectors (the ISF parameter vectors causing TH_{med} to be exceeded the most) are replaced by the median ISF parameter vector prior to computing the averaged ISF parameter vector $\mathbf{f}^{mean}$.

The set of ISF parameter vectors obtained as a result of the median replacement are denoted as $\mathbf{f'}(n - i)$, where n is the index of the current frame, and i is the averaging period index (i=0,...,7).

When the median replacement is performed at the end of the hangover period (first CN update), all of the ISF parameter vectors $\mathbf{f}(n - i)$ of the 7 previous frames (the hangover period, i=1,...,7) have quantized values, while the ISF parameter vector $\mathbf{f}(n)$ at the most recent frame n has unquantized values. In the subsequent CN updates, the ISF parameter vectors of the CN averaging period in the frames overlapping with the hangover period have quantized values, while the parameter vectors of the more recent frames of the CN averaging period have unquantized values. When the period of the eight most recent frames is non-overlapping with the hangover period, the median replacement of ISF parameters is performed using only unquantized parameter values.

The averaged ISF parameter vector $\mathbf{f}^{mean}(n)$ at frame n shall be computed according to the equation:

$$\mathbf{f}^{mean}(n) = \frac{1}{8}\sum_{i=0}^{7} f'(n-i), \tag{A-4}$$

where $\mathbf{f'}(n - i)$ is the ISF parameter vector of one of the eight most recent frames (i=0,...,7) after performing the median replacement, i is the averaging period index, and n is the frame index.

The averaged ISF parameter vector $\mathbf{f}^{mean}(n)$ at frame n is quantized using the comfort noise ISF quantization tables The mean removed ISF vector to be quantized is obtained according to the following equation:

$$r(n) = \mathbf{f}^{mean}(n) - \bar{\mathbf{f}} \tag{A-5}$$

where $\mathbf{f}^{mean}(n)$ is the averaged ISF parameter vector at frame n, $\bar{\mathbf{f}}$ is the constant mean ISF vector, $r(n)$ *is* the computed ISF mean removed vector at frame n, and n is the frame index.

A.4.2 Frame energy calculation

The frame energy is computed for each frame marked with VAD = 0 according to the equation:

$$en_{\log}(i) = \frac{1}{2}\log_2\left(\frac{1}{N}\sum_{n=0}^{N-1} s^2(n)\right) \tag{A-6}$$

where $s(n)$ is the high-pass-filtered input speech signal of the current frame i. The energy is also adjusted according to the signalled speech modes capabilities, as to provide high quality transitions from Comfort Noise to Speech.

The averaged logarithmic energy is computed by:

$$en_{\log}^{mean}(i) = \frac{1}{8}\sum_{n=0}^{7} en_{\log}(i-n) \tag{A-7}$$

The averaged logarithmic energy is quantized using a 6-bit arithmetic quantizer. The 6-bits for the energy index are transmitted in the SID frame (see bit allocation in Table A.1).

Table A.1/G.722.2 – Source encoder output parameters in order of occurrence and bit allocation for comfort noise encoding

Bits (MSB-LSB)	Description
s1-s6	Index of 1st ISF subvector
s7-s12	Index of 2st ISF subvector
s13-s18	Index of 3nd ISF subvector
s19-s23	Index of 4th ISF subvector
s24-s28	Index of 5th ISF subvector
s29-s34	Index of logarithmic frame energy
s35	Dithering flag

A.4.3 Analysis of the variation and stationarity of the background noise

The encoder first determines how stationary background noise is. Dithering is employed for non-stationary background noise. The information about whether to use dithering or not is transmitted to the decoder using a binary information (CN_{dith} – flag).

The binary value for the CN_{dith} – flag is found by using the spectral distance ΔS_i of the spectral parameter vector $\mathbf{f}(i)$ to the spectral parameter vectors $\mathbf{f}(j)$ of all the other frames j=0,..., l_{dtx}-1, $j \neq i$ within the CN averaging period (l_{dtx}). The computation of the spectral distance is described in A.4.1. A sum of spectral distances $D_s = \sum_{i=0}^{7} \Delta S_i$ is then computed. If D_S is small, CN_{dith} – flag is set to 0. Otherwise, CN_{dith} – flag is set to 1. Additionally, variation of energy between frames is studied. The sum of absolute deviation of $en_{log}(i)$ from the average en_{log} is computed. If the sum is large, CN_{dith} – flag is set to 1, even if the flag was earlier set to 0.

A.4.4 Modification of the speech encoding algorithm during SID frame generation

When the TX_TYPE is not equal to SPEECH, the speech encoding algorithm is modified in the following way:

– The non-averaged LP parameters which are used to derive the filter coefficients of the filters $H(z)$ and $W(z)$ of the speech encoder are not quantized;

– The open loop pitch lag search is performed, but the closed loop pitch lag search is inactivated. The adaptive codebook memory is set to zero.

– No fixed codebook search is made.

– The memory of weighting filter $W(z)$ is set to zero, i.e. the memory of $W(z)$ is not updated.

– The ordinary LP parameter quantization algorithm is inactive. The averaged ISF parameter vector $\mathbf{f}^{mean}$ is calculated each time a new SID frame is to be sent. This parameter vector is encoded into the SID frame as defined in A.4.1.

– The ordinary gain quantization algorithm is inactive.

– The predictor memories of the ordinary LP parameter quantization algorithm is initialized when TX_TYPE is not SPEECH, so that the quantizers start from known initial states when the speech activity begins again.

In the 23.85 kbit/s mode, when the TX_TYPE is equal to SPEECH and VAD is OFF, the speech encoding algorithm is modified in the following way:

– The generation of high-band gain g_{HB} is changed by adapting it during non-active speech period towards estimated gain in order to ensure smooth transition of high-band gain. g_{HB} is then:

$$g_{HB} = \frac{hang_{DTX}}{7} g_{HB} + (1 - \frac{hang_{DTX}}{7}) g_{est} \tag{A-8}$$

where $hang_{DTX}$ is DTX counter.

A.4.5 SID-frame encoding

The encoding of the comfort noise bits in a SID frame is described in Annex E/G.722.2 where the indication of the first SID frame is also described. The bit allocation and sequence of the bits from comfort noise encoding is shown in Table A.1.

A.5 Functions on the receive (RX) side

The situations in which comfort noise shall be generated on the receive side are defined in Annex B/G.722.2. In general, the comfort noise generation is started or updated whenever a valid SID frame is received.

A.5.1 Averaging and decoding of the LP and energy parameters

When speech frames are received by the decoder the LP and the energy parameters of the last seven speech frames shall be kept in memory. The decoder counts the number of frames elapsed since the last SID frame was updated and passed to the RSS by the encoder. Based on this count, the decoder determines whether or not there is a hangover period at the end of the speech burst (defined in Annex B/G.722.2). The interpolation factor is also adapted to the SID update rate.

As soon as a SID frame is received, comfort noise is generated at the decoder end. The first SID frame parameters are not received but computed from the parameters stored during the hangover period. If no hangover period is detected, the parameters from the previous SID update are used.

The averaging procedure for obtaining the comfort noise parameters for the first SID frame is as follows:

– When a speech frame is received, the ISF vector is decoded and stored in memory; moreover the logarithmic frame energy of the decoded signal is also stored in memory.

– The averaged values of the quantized ISF vectors and the averaged logarithmic frame energy of the decoded frames are computed and used for comfort noise generation.

The averaged value of the ISF vector for the first SID frame is given by:

$$\hat{\mathbf{f}}^{mean}(i) = \frac{1}{8}\sum_{n=0}^{7}\hat{\mathbf{f}}(i-n) \tag{A-9}$$

where $\hat{\mathbf{f}}(i-n)$, $n > 0$ is the quantized ISF vector of one of the frames of the hangover period and where $\hat{\mathbf{f}}(i-0) = \hat{\mathbf{f}}(i-1)$. The averaged logarithmic frame energy for the first SID frame is given by:

$$\hat{en}_{\log}^{mean}(i) = \frac{1}{8}\sum_{n=0}^{7}\hat{en}_{\log}(i-n) \tag{A-10}$$

where $\hat{en}_{\log}(i-n)$, $n > 0$ is the logarithmic vector of one of the frames of the hangover period computed for the decoded frames and where $\hat{en}_{\log}(i-0) = \hat{en}_{\log}(i-1)$.

For ordinary SID frames, the ISF vector and logarithmic frame energy are computed by table lookup. The ISF vector is given by the sum of the decoded reference vector and the constant mean ISF vector.

During comfort noise generation the spectrum and energy of the comfort noise is determined by interpolation between old and new SID frames.

When dithering is used, the ISF vector **f** is modified by:

$$\mathbf{f}(i) = \mathbf{f}(i) + rand(-L(i), L(i)) \qquad i = 1,...,16 \tag{A-11}$$

where $L(i) = 100 + 0.8i$ Hz and $rand(-L(i),L(i))$ is random function generating values between $-L(i)$ and $L(i)$. A minimum gap of 175 Hz is ensured between elements of **f**.

Dithering insertion for energy parameter is similar to spectral dithering and can be computed as follows:

$$en_{\log}^{mean} = en_{\log}^{mean} + rand(-L, L) \tag{A-12}$$

where $L = 75$ and $en_{\log}^{mean}$ is the energy value used for scaling the energy of the comfort noise excitation.

A.5.2 Comfort noise generation and updating

The comfort noise generation procedure uses the Adaptive Multi-Rate Wideband (AMR-WB) speech decoder algorithm defined in the main body of ITU-T Rec. G.722.2.

When comfort noise is to be generated, the various encoded parameters are set as follows:

In each subframe, the pulse positions and signs of the excitation are locally generated using uniformly distributed pseudo-random numbers. The excitation pulses take values between +2047 and –2048 when comfort noise is generated. The fixed codebook comfort noise excitation generation algorithm works as follows:

for (i = 0; i < 64; i++) u[i] = shr(random(),4)

where:

u[0..63]	excitation buffer;
random()	generates a random integer value, uniformly distributed between –32 768 and +32 767;

The excitation gain is computed from the logarithmic frame energy parameter by converting it to the linear domain.

The adaptive codebook gain values in each subframe are set to 0; also, the memory of the adaptive codebook is set to zero.

The pitch delay values in each subframe are set to 64.

The LP filter parameters used are those received in the SID frame.

The predictor memory of the ordinary LP parameter algorithm is initialized when RX_TYPE is not SPEECH , so that the quantizer start from given initial states when the speech activity begins again. With these parameters, the speech decoder now performs the standard operations described in the main body of ITU-T Rec. G.722.2 and synthesizes comfort noise. During CN generation, the high-band generation is performed using estimated high-band gain like in 8.85, 12.65, 14.25, 15.85, 18.25, 19.85 or 23.05 kbit/s modes during active speech.

Updating of the comfort noise parameters (energy and LP filter parameters) occurs each time a valid SID frame is received, as described in Annex B/G.722.2.

When updating the comfort noise, the parameters above should be interpolated over the SID update period to obtain smooth transitions.

A.6 Computational details and bit allocation

A bit exact computational description of comfort noise encoding and generation in form of an ANSI-C source code is found in Annex C/G.722.2.

The detailed bit allocation and the sequence of bits in the comfort noise encoding is shown in Table A.1.

SERIES OF ITU-T RECOMMENDATIONS

Series A	Organization of the work of ITU-T
Series B	Means of expression: definitions, symbols, classification
Series C	General telecommunication statistics
Series D	General tariff principles
Series E	Overall network operation, telephone service, service operation and human factors
Series F	Non-telephone telecommunication services
Series G	**Transmission systems and media, digital systems and networks**
Series H	Audiovisual and multimedia systems
Series I	Integrated services digital network
Series J	Cable networks and transmission of television, sound programme and other multimedia signals
Series K	Protection against interference
Series L	Construction, installation and protection of cables and other elements of outside plant
Series M	TMN and network maintenance: international transmission systems, telephone circuits, telegraphy, facsimile and leased circuits
Series N	Maintenance: international sound programme and television transmission circuits
Series O	Specifications of measuring equipment
Series P	Telephone transmission quality, telephone installations, local line networks
Series Q	Switching and signalling
Series R	Telegraph transmission
Series S	Telegraph services terminal equipment
Series T	Terminals for telematic services
Series U	Telegraph switching
Series V	Data communication over the telephone network
Series X	Data networks and open system communications
Series Y	Global information infrastructure and Internet protocol aspects
Series Z	Languages and general software aspects for telecommunication systems

Geneva, 2002