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

International telephone connections and circuits – General Recommendations on the transmission quality for an entire international telephone connection

One-way transmission time

ITU-T Recommendation G.114

(Formerly CCITT Recommendation)

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ITU-T Recommendation G.114

One-way transmission time

Summary

This ITU-T Recommendation provides specifications for transmission time, including delay due to equipment processing time as well as propagation delay, in connections with echo adequately controlled. Recognizing that delay became a limited resource in modern networks, this ITU-T Recommendation is intended to assist network operators and transmission planners as well as equipment manufacturers in controlling the detrimental effects of pure delay (even if echo is perfectly controlled) on end-to-end speech transmission quality. All applications with overall performance which depend on user or terminal interactivity are concerned.

Source

ITU-T Recommendation G.114 was revised by ITU-T Study Group 12 (1997-2000) and approved under the WTSC Resolution 1 procedure on 18 May 2000.

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 Conference (WTSC), 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 WTSC 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.

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As of the date of approval of this Recommendation, ITU had not 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

Introduction

Transmission time for connections with digital segments includes delay due to equipment processing as well as propagation delay, such that both types of delay can be significant contributors to overall transmission time. Guidance is especially needed for designers of telecommunications equipment that uses signal processing, causing increase in delay; including delays resulting from packetization, as performed by ATM, Frame Relay, Voice over Internet Protocol (VoIP), etc. systems, as well as from codec-related processing delays.

Historically a value of 400 ms was considered a meaningful limit for network planning purposes, where speech transmission performance was the focus. This value was not originally intended as guidance for equipment designers who, on an increasingly frequent basis, can substantially affect the transmission time by the amount of signal processing in their designs.

Transmission time is a very important parameter for any application whose overall performance is dependent on user or terminal interactivity. Applications such as voice, voiceband data, digital data, and videotelephony may involve user tasks or terminal equipment characteristics that vary substantially in their sensitivity to transmission delay. Because network and service providers cannot alter the transmission time characteristics nor transmission media between two Administrations, in response to all possible user tasks and applications, some highly interactive tasks may experience degradation even at delays on the order of 100 ms. Accordingly, it is critical that the delay (transmission time) be seen as a vital resource that is to be consumed with caution, and only when clear service benefits derive from it. This especially applies to delay associated with signal processing.

This ITU-T Recommendation is intended to assist equipment designers and network planners in realizing acceptable services to users performing a wide variety of tasks with multiple applications. It is recognized that not all possible user applications and network configurations can be predicted, and that some user applications and network arrangements may combine processing delays and propagation times such that the total transmission time exceeds 400 ms.

A clear purpose of this ITU-T Recommendation is thus to emphasize the need to consider the delay impact on evolving telecommunication applications, and indicate the desirability of avoiding delay increases, especially processing delays, whenever possible.

ITU-T Recommendation G.114

One-way transmission time

1 Introduction

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2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent edition of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

[1] CCITT Recommendation E.171 (1988), *International telephone routing plan*.

[2] ITU-T Recommendation G.100 (1993), *Definitions used in Recommendations on general characteristics of international telephone connections and circuits.*

[3] ITU-T Recommendation G.103 (1998), *Hypothetical reference connections*.

[4] ITU-T Recommendation G.131 (1996), *Control of talker echo*.

[5] ITU-T Recommendation G.165 (1993), *Echo cancellers*.

[6] ITU-T Recommendation G.763 (1998), *Digital circuit multiplication equipment using G.726 ADPCM and digital speech interpolation*.

[7] CCITT Recommendation G.764 (1990), *Voice packetization – Packetized voice protocols*.

[8] ITU-T Recommendation G.766 (1996), *Facsimile demodulation/remodulation for digital circuit multiplication equipment.*

[9] ITU-T Recommendation G.767 (1998), *Digital circuit multiplication equipment using 16 kbit/s LD-CELP, digital speech interpolation and facsimile demodulation/remodulation.*

[10] CCITT Recommendation G.801 (1988), *Digital transmission models*.

[11] CCITT Recommendation P.82 (1988), *Method for evaluation of service from the standpoint of speech transmission quality*.

[12] ITU-T Recommendation Q.551 (1996), *Transmission characteristics of digital exchanges*.

3 End-to-end transmission time limits

In consideration of the above points, the ITU-T *recommends* the following limits for one-way transmission time for connections with echo adequately controlled (see Note 1) according to ITU-T Recommendation G.131 [4]:

– 0 to 150 ms: Acceptable for most user applications (see Note 2).

– 150 to 400 ms: Acceptable provided that Administrations are aware of the transmission time impact on the transmission quality of user applications (see Note 3).

– above 400 ms: Unacceptable for general network planning purposes; however, it is recognized that in some exceptional cases (see Note 4) this limit will be exceeded.

NOTE 1 – The use of echo control equipment that introduces other impairments, such as speech clipping and noise contrast, may have to be controlled in order to achieve acceptable transmission quality.

NOTE 2 – Some highly interactive voice and data applications may experience degradation for values below 150 ms. Therefore, increases in processing delay on connections with transmission times even well below 150 ms should be discouraged unless there are clear service and application benefits.

NOTE 3 – For example, international connections with satellite hops that have transmission times below 400 ms are considered acceptable.

NOTE 4 – Examples of such exceptions are unavoidable double satellite hops, satellites used to restore terrestrial routes, fixed satellite and digital cellular interconnections, videotelephony over satellite circuits, and very long international connections with two digital cellular systems connected by long terrestrial facilities.

The recommended limits given here can be better interpreted if the information provided in Annex B is considered. For example, the voice quality test results indicate that, even in the complete absence of echo, 10% or more of the speakers may experience difficulty due to a delay of 400 ms. Increases in delay beyond this value will cause a further increase in unacceptable connections, especially for highly interactive conversations. To provision services with route diversity and restoration capabilities, Administrations may nonetheless choose to exceed 400 ms, on an exceptional basis. The data in Annex A provides guidance as to the service quality impact of such a decision.

4 Transmission time allocation

As transmission time becomes more of a limited resource in the modern digital networks, it is important to make an effort to minimize its increase caused by the introduction of new delay-prone technologies.

The delay allocation rules, recommended here, apply to the processing time only and do not include the propagation time portion of the total connection delay. The propagation time is determined by the distance and the speed of the signal in the transmission facility and can be controlled only in a very limited way by the network planners. The key factor is the geographical distance that differs widely within and between various countries. Also, in practice, the routing choices in international and national networks are often made for other than performance related reasons (e.g. economics, traffic considerations) with the facilities such as satellites or radio links, commonly used for routing diversity, representing large investment value that cannot be easily replaced.

In consideration of the above points, no more than 50 ms of one-way processing time is recommended for each of the national systems and for the international chain of circuits of an international connection (terrestrial or satellite) carrying speech signals. While the total processing time in any of these three parts of the international connection should be kept below 50 ms, the processing time is usually much less than this. For example, for the typical connection given in ITU-T Recommendation G.801 [10] (see Figure 2/G.801 [10]), total processing time associated with switches, cross connects, multiplexers, etc., should be around 6 ms for a national system and 3 ms for the international chain. It should be noted that the recommended guidelines may be exceeded with today's technology, if a scenario is encountered which includes low bit-rate coders (e.g. a circuit with low bit-rate coding according to ITU-T Recommendation G.729A at 8 kbit/s).

ANNEX A

Delay estimation for circuits

In the establishment of the general interconnection plan within the limits in this Recommendation, the one-way transmission time of both the national extension circuits and the international circuits must be taken into account. The transmission time of circuits and connections is the aggregate of several components: e.g. group delay in cables and equipment processing times (e.g. digital switches, low bit-rate coders), etc.

The general planning values given in Table A.1 and those provided in Appendix I for delay incurred due to coder-related processing may be used to estimate the total transmission time of specified assemblies which may form circuits or connections.

Table A.1/G.114

Transmission or processing system	Contribution to one-way transmission time	Remarks
Terrestrial coaxial cable or radio-relay system: FDM and digital transmission	4 μs/km	Allows for delay in repeaters and regenerators
Optical fibre cable system, digital transmission	5 μs/km (Note 1)	
Submarine coaxial cable system	6 μs/km	
Submarine optical fibre system: – transmit terminal – receive terminal	 13 ms 10 ms	Worst case
Satellite system: – 400 km altitude – 14 000 km altitude – 36 000 km altitude	 12 ms 110 ms 260 ms	Propagation through space only (between earth stations)
FDM channel modulator or demodulator	0.75 ms (Note 2)	
PLMS (Public Land Mobile System) – objective 40 ms	80-110 ms	

Table A.1/G.114 ***(concluded)***

Transmission or processing system	Contribution to one-way transmission time	Remarks
H.260-series video coders and decoders	Further study (Note 3)	Half the sum of transmission times in both directions of transmission
DCME (ITU-T Rec. G.763 [6]) per pair: for speech, VBD, and non-remodulated fax	30 ms	
DCME (ITU-T Rec. G.767) per pair: for speech, VBD, and non-remodulated fax	30 ms	
DCME (ITU-T Rec. G.766 [8] in conjunction with ITU-T Rec. G.763 [6] or ITU-T Rec. G.767 [9]) per pair: for remodulated fax	200 ms	
PCME (Rec. G.764 [7]) per pair: – with speech and non-remodulated VBD – with remodulated VBD	 35 ms 70 ms	
Transmultiplexer	1.5 ms (Note 4)	
Digital transit exchange, digital-digital	0.45 ms (Note 5)	
Digital local exchange, analogue-analogue	1.5 ms (Note 5)	
Digital local exchange, analogue subscriber line-digital junction	0.975 ms (Note 5)	
Digital local exchange, digital subscriber line-digital junction	0.825 ms (Note 5)	
Echo cancellers	0.5 ms (Note 6)	
ATM (CBR using AAL1)	6.0 ms (Note 7)	

NOTE 1 – This value is provisional and is under study.

NOTE 2 – These values allow for group-delay distortion around frequencies of peak speech energy and for delay of intermediate higher order multiplex and through-connecting equipment.

NOTE 3 – Further study required. Delay for these devices is usually non-constant, and the range varies by implementation. Current implementations are on the order of several hundred milliseconds and considerable delay is added to audio channels to achieve lip-synchronization. Manufacturers are encouraged to reduce their contribution to transmission time, in accordance with this ITU-T Recommendation.

NOTE 4 – For satellite digital communications where the transmultiplexer is located at the earth station, this value may be increased to 3.3 ms.

NOTE 5 – These are mean values: depending on traffic loading, higher values can be encountered, e.g. 0.75 ms (1.950 ms, 1.350 ms, or 1.250 ms) with 0.95 probability of not exceeding. (For details see ITU-T Recommendation Q.551 [12].)

NOTE 6 – This is averaged for both directions of transmission.

NOTE 7 – This is the cell formation delay of 64 kbit/s stream when it completely fills the cell (one voice channel per VC). In practical applications, additional delay will result, e.g. from cell loss detection and buffering. Other delays may be applicable to other AALs and cell mapping arrangements, and are for further study.

A.1 Planning values of transmission time

Provisionally, the planning values of transmission time in Table A.1 together with those related to coders in Appendix I may be used.

A.2 National extension circuits

The main arteries of the national network should consist of high bit-rate, low latency propagation lines which normally do not introduce additional low bit-rate coders. In these conditions, the transmission time between the international centre and the subscriber farthest away from it in the national network can be estimated as follows:

a) In purely analogue networks, the transmission time will probably not exceed:

$$12 + (0.004 \times \text{distance in kilometres}) \text{ ms}$$

Here the factor 0.004 is based on the assumption that national trunk circuits will be routed over high-velocity plant (250 km/ms). The 12 ms constant term makes allowance for terminal equipment and for the probable presence in the national network of a certain quantity of loaded cables (e.g. three pairs of channel translating equipments plus about 160 km of H 88/36 loaded cables). For an average size country (see Figure 5/G.103 [3]) the one-way propagation time will be less than 18 ms.

b) In mixed analogue/digital networks, the transmission time can generally be estimated by the equation given for purely analogue networks. However, under certain unfavourable conditions, increased delay may occur compared with the purely analogue case. This occurs in particular when digital exchanges are connected with analogue transmission systems through PCM/FDM equipment in tandem, or transmultiplexers. With the growing degree of digitalization, the transmission time will gradually approach the condition of purely digital networks.

c) In purely digital networks between local exchanges, based on optical fibre systems (e.g. an IDN), the transmission time will probably not exceed:

$$3 + (0.005 \times \text{distance in kilometres}) \text{ ms}$$

The 3 ms constant term makes allowance for one pair of PCM coder and decoder and for five digitally switched exchanges.

NOTE – The value 0.005 is a mean value for optical fibre systems; for coaxial cable systems and radio-relay systems 0.004 is to be used.

d) In purely digital networks between subscribers (e.g. an ISDN), the delay of c) above has to be increased by up to 3.6 ms if burst-mode (time compression multiplexing) transmission is used on 2-W local subscriber lines.

These values do not cover the additional delays introduced by PBXs and Private Branch Networks (PBNs).

A.3 International circuits

International circuits will use high-velocity transmission systems, e.g. terrestrial cable or radio-relay systems, submarine systems or satellite systems. The planning values of Table A.1 may be used; for delay introduced due to codec-related processing the values provided in Appendix I may be used.

The magnitude of the mean one-way transmission time for circuits on high altitude communication satellite systems makes it desirable to impose some routing restrictions on their use. Details of these restrictions are given in ITU-T Recommendation E.171 [1].

A.4 Codec delay

Modern speech codecs operate on collections of speech samples known as frames. Each block of input speech samples is processed into a compressed frame. The coded speech frame is not generated until all speech samples in the input block have been collected by the encoder. Thus, there is a delay of one frame before processing can begin. In addition many coders also look into the succeeding frame to improve compression efficiency. The length of this advance look is known as the look-ahead time of the coder. The time required to process an input frame is assumed to be the same as the frame length since efficient use of processor resources will be accomplished when an encoder/decoder pair (or multiple encoder/decoder pairs operating in parallel on multiple input streams) fully uses the available processing power (evenly distributed in the time domain). Thus, the delay through an encoder/decoder pair is normally assumed to be:

$$2 \times \text{frame size} + \text{look-ahead}$$

A.4.1 Delay in wirebound environment

If the output facility is running at the same rate as the speech codec (e.g. an 8 kbit/s facility for ITU-T G.729), then an additional frame of delay is incurred when clocking the compressed frame to the facility. Thus, the maximum delay attributable to codec-related processing in conventional wirebound systems (i.e. the PSTN) is:

$$3 \times \text{frame size} + \text{look-ahead}$$

A.4.2 Delay in mobile and wireless environment

If the output facility is a mobile network or a cordless facility, then the frame output by the encoder will function similar to the operation in wirebound environment but an additional delay is incurred for attaching the compressed frame to the airpath (assumed again that the mobile facility is running at the same rate as the speech codec). Thus, the maximum delay attributable to codec-related processing in mobile and wireless systems is:

$$3 \times \text{frame size} + \text{look-ahead} + \text{air interface framing}$$

A.4.3 Delay in IP environment (one frame per packet)

If the output facility is an IP network, then the frame output by the encoder will instantaneously be dropped into an IP packet. The additional delay required for IP packet assembly and presentation to the underlying link layer will depend on the link layer. When the link layer is a LAN (e.g. Ethernet), this additional time will usually be quite small. Thus, the minimum delay attributable to codec-related processing in IP-based systems is:

$$2 \times \text{frame size} + \text{look-ahead}$$

When the link layer is one with lower clock rate (e.g. Modem connection) or one with high traffic load (e.g. congested LAN), the additional delay will increase substantially. In order to clock compressed frames at least with the same rate to the facility as the speech samples are collected at the input of the encoder, the additional delay should not exceed one frame size. Thus, the maximum delay attributable to codec-related processing in IP-based systems operating in real-time is:

$$3 \times \text{frame size} + \text{look-ahead}$$

A.4.4 Delay in IP environment (multiple frames per packet)

If multiple voice frames are grouped together into a single IP packet, further delay is added to the speech signal. This delay will be at least the duration of one extra voice frame at the encoder for each additional voice frame added to the IP packet. Thus, the minimum delay attributable to codec-related processing in IP-based systems with multiple frames per packet is:

$$(N + 1) \times \text{frame size} + \text{look-ahead}$$

where N is the number of frames in each packet.

When the link layer is one with lower clock rate (e.g. Modem connection) or one with high traffic load (e.g. congested LAN), additional delay will be incurred in delivering the packet to the facility. In order to clock compressed frames at least with the same rate to the facility as the speech samples are collected at the input of the encoder, the additional delay should, in case of multiple frames per packet, not exceed the length of the frames contained in one packet. It should be noted that clocking out a packet to the IP facility cannot start before all speech frames for this packet are available. Thus, the maximum delay attributable to codec-related processing in IP-based systems operating in real-time with multiple frames per packet is:

$$(2N + 1) \times \text{frame size} + \text{look-ahead}$$

where N is the number of frames in each packet.

The following Figure A.1 provides an example for N = 2:

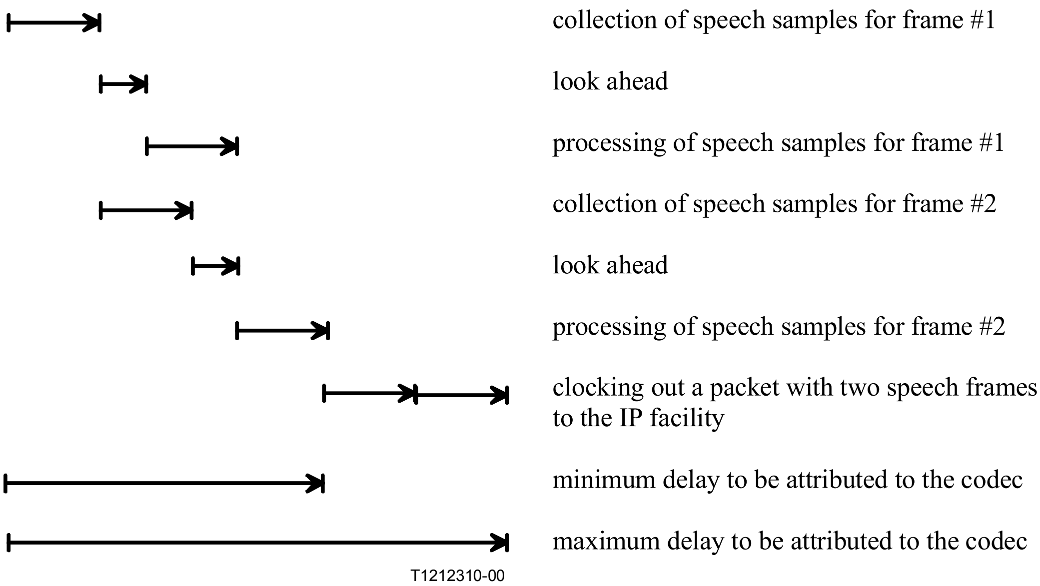


Figure A.1/G.114 – Example: Composition of total codec-related delay in an IP Environment for N = 2

A.5 Delay due to IP delay variation buffer

Packetized transmission systems exhibit variable delay in packet delivery time; this is caused by the fact that different packets carrying speech samples of the same telephone conversation may be transported via distinct routes through the network. Details of this effect depend strongly on the specific mechanisms for transport, queuing or prioritization, which may be implemented in such a system.

Packets which have been transported through a packet-based network are collected in a buffer at the receive side. This buffer functions as the instance which re-arranges the timely order of the packets. If the delivery time of a packet exceeds the length of the receive buffer, then this packet "comes too late" with respect to the size of this buffer and will be discarded. Hence, the speech carried in this packet is lost for the decoding process. This "packet loss" impacts speech transmission quality. One approach to minimize the percentage of such lost packets is the dynamic adaptation of the length of the receive buffer.

A receive buffer with adaptive length controls its actual length via its filling level:

- if the number of packets being at a time within the buffer increases, then the length of the buffer will be increased and at the same time at the output of the buffer short pause sequences from the original signal will be dropped in order to drain the buffer with a faster rate;

- if the number of packets being at a time within the buffer decreases, then the length of the buffer will be decreased and at the same time at the output of the buffer short pause sequences will be additionally inserted into the original signal in order to drain the buffer with a slower rate.

Consequently, there will be a variable value of the end-to-end mean one-way delay between the talker's mouth and the listener's ear. It is important to clearly distinguish this effect from other delay variation discussions which refer to network internal processes only. The impact of an end-to-end delay variation – as explained in this subclause – is strongly dependent on the length of the dropped or inserted pieces of pauses; further emphasis rests on the correct implementation of the dynamic adaptation processes, e.g. the insertion of pieces of pauses into a syllable will have more serious impact than the insertion of pieces of pauses into a pause sequence.

ANNEX B

Long delay considerations for telephone, videotelephone and videoconference circuits

B.1 Introduction

International connections (see Figure 1/G.103 [3]) comprising submarine cables may involve a maximum one-way transmission delay of about 170 ms.

A one-hop satellite connection even with an ISL (Inter-Satellite Link) of moderate length introduces one-way transmission delay within the recommended limit of 400 ms. However, a careful analysis of the additional probable delay contributions by digital signal processing (e.g. TDMA, DSI, DCME, 16 kbit/s, 32 kbit/s and lower bit rate encoding, bit-regeneration, packet-switching, etc.), among other sources, shows that in some cases the recommended limit of 400 ms mean one-way transmission time might be exceeded.

In light of recent technical improvements in echo-control techniques and considering that fixed processing delays may reach hundreds of milliseconds in some currently designed systems (e.g. low bit rate digital mobile systems), it is important to understand also the effects of delay, in the absence of echo, on communications. This annex addresses this issue.

The 4-wire circuits provide a close approximation to echo-free connections, assuming adequate acoustic coupling loss across the handset. In the long run, with expansion of the ISDN implementation, use of 4-wire circuits is expected to grow. However, 2-wire circuits and their accompanying hybrid connection, as well as other components causing echo, are still likely to be present in varying degrees during the foreseeable future. Thus, the use of modern echo cancellers in satellite circuits is currently regarded as the most effective method for overcoming the echo problem, provided that the characteristics of the echo path to be modelled by the echo canceller are linear and time invariant, or varying only slowly compared with the convergence speed of the echo canceller.

A brief discussion of delay effect, in the absence of echo, on communication quality is provided below.

B.2 Effect of long transmission delays on the subscriber

B.2.1 Effects of echo cancellers

In 1987, a series of tests was performed to determine the effectiveness of echo cancellers in terrestrial and satellite circuits, using echo cancellers conforming to ITU-T Recommendation G.165 [5] and a callback interview procedure as per Annex A/P.82 [11]. A summary of the results is shown in Figure B.1, giving a plot of the per cent difficulty as a function of one-way transmission time. A one-way delay value of 45 ms over terrestrial circuits was taken as a reference, and the effect of increasing the delay value to 300 ms and 500 ms over terrestrial and satellite links was evaluated.

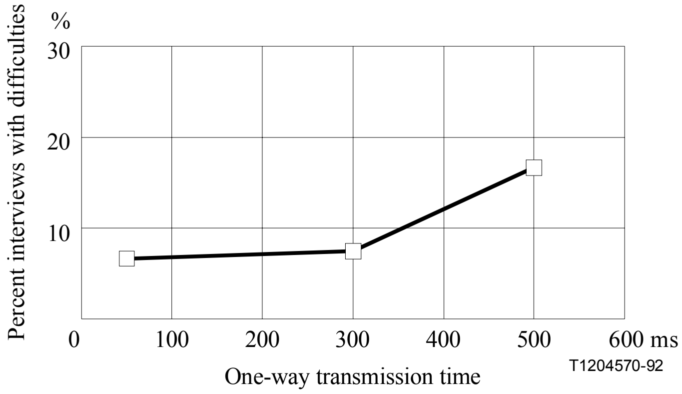


Figure B.1/G.114 – Effect of long one-way transmission times on the difficulty of conversation with echo cancellers in the circuit

These results show that no significant difference between 45 ms and 300 ms delays resulted for the "per cent difficulty" score. At a 500 ms delay, the per cent difficulty score approximately doubled (from 7.3% to 15.8%), but this value is still considerably smaller than earlier results of over 60% obtained with echo suppressors.

The above results support the view that connections with delays somewhat greater than 400 ms may be accepted provided that echo cancellers conforming to the specifications of ITU-T Recommendation G.165 [5] or other echo control devices with equivalent performance are used.

B.2.2 Effects of delay on dynamics of conversation

The most recent evidence presented by some Administrations suggests that the performance degradation due to conversation dynamics impairments is noticeable even below 400 ms one-way delay limit. This effect can be observed when structured interactive tasks and selected sensitive measures are employed in subjective experimentation.

In 1989, a series of subjective experiments was conducted to determine the impact of the delay on the conversational characteristics deemed to be important in a business-type environment. A structured conversational task coupled with objective and subjective measures of the temporal dynamics of the conversation were developed and used in the experiment. Subjective measures included ratings on the ease of interruption, the necessity of repeating utterances, the attentiveness, responsiveness and helpfulness of the partner. The results are shown in Figure B.2.

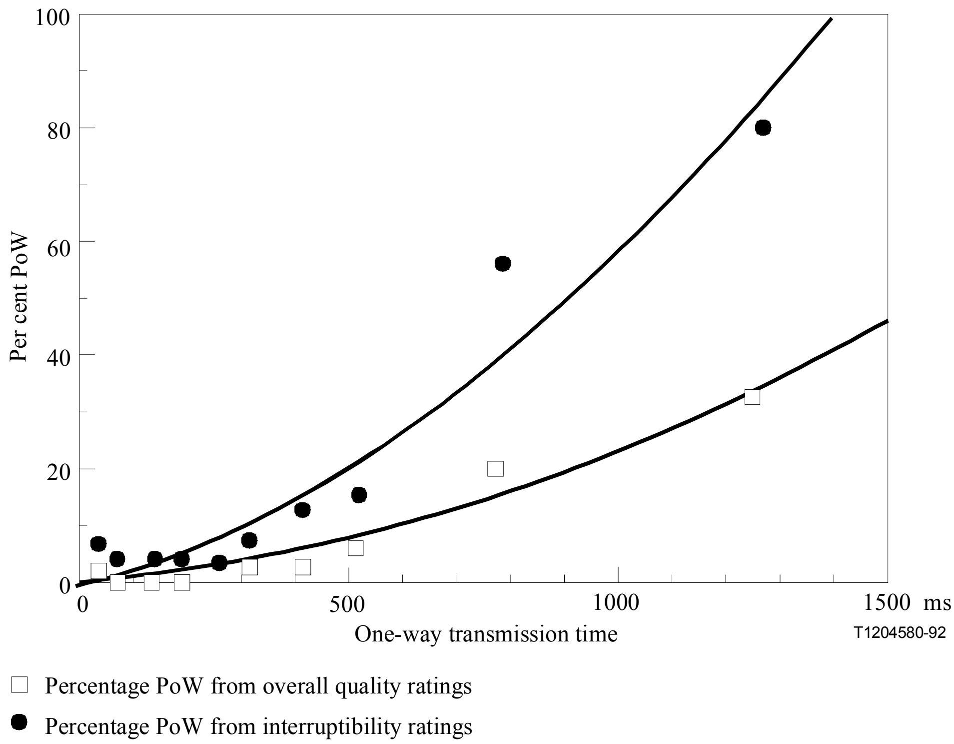


Figure B.2/G.114 – Comparison of PoW for overall quality and interruptability

A subjective test intended to evaluate the effects of pure delay on speech quality was completed in 1990. The test was designed to obtain subjective reactions, in the context of interruptability and quality, to echo-free telephone circuits in which various amounts of delay were introduced. The results indicated that long delays did not greatly reduce mean opinion scores over the range of delay tested, viz. 0 to 1000 ms of one-way delay. In addition, the measure of interruptability did not show the divergence from overall quality to be as significant as indicated in Figure B.2. However, observations during the test and subject interviews after the test showed the subjects experienced some real difficulties in communicating at the longer delays, although subjects did not always associate the difficulty with the delay.

A second subjective test intended to evaluate the effects of pure delay on telephone connections used by volunteer Telco customers was completed in 1991. The calls from these customers were routed through a laboratory where varying amounts of delay, viz. 0 to 750 ms of one-way delay was added. The test results showed that calls with (one-way delay): 0 ms of inserted delay were rated "good"; 250 ms of inserted delay were rated "fair"; and 500 ms of inserted delay were rated "poor". These results are presented in Figure B.3.

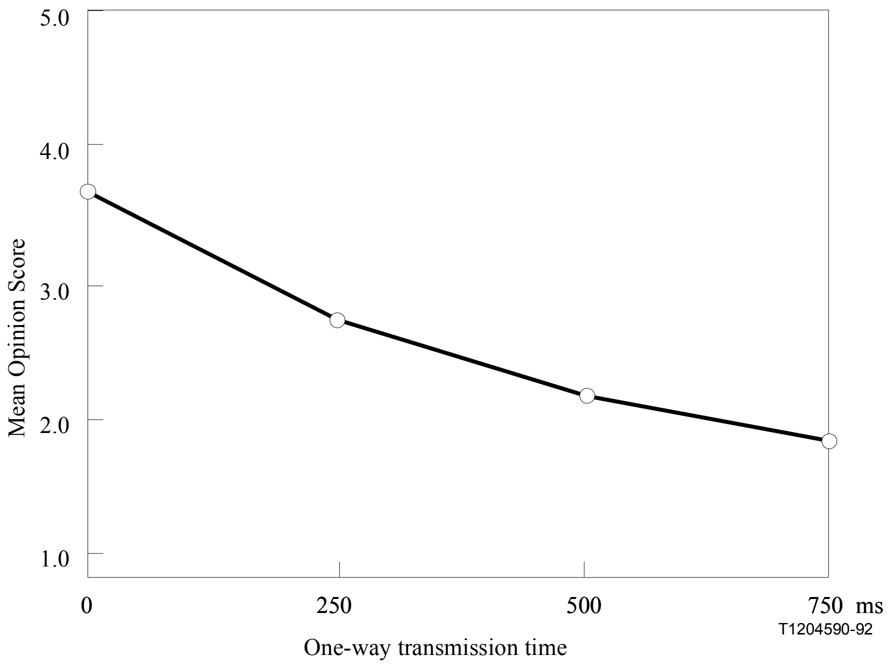


Figure B.3/G.114 – Mean Opinion Scores (MOS) for the four delay conditions

Similar experiments were conducted during the same time frame by various Administrations. The following is a highlight of the results presented.

The effect of delay was measured using a combination of objective physical parameters related to efficiency of a conversation. It was studied using the following six different conversational modes (tasks):

– Task 1: Read out random numbers as quickly as possible in turn.

– Task 2: Verify random numbers as quickly as possible in turn.

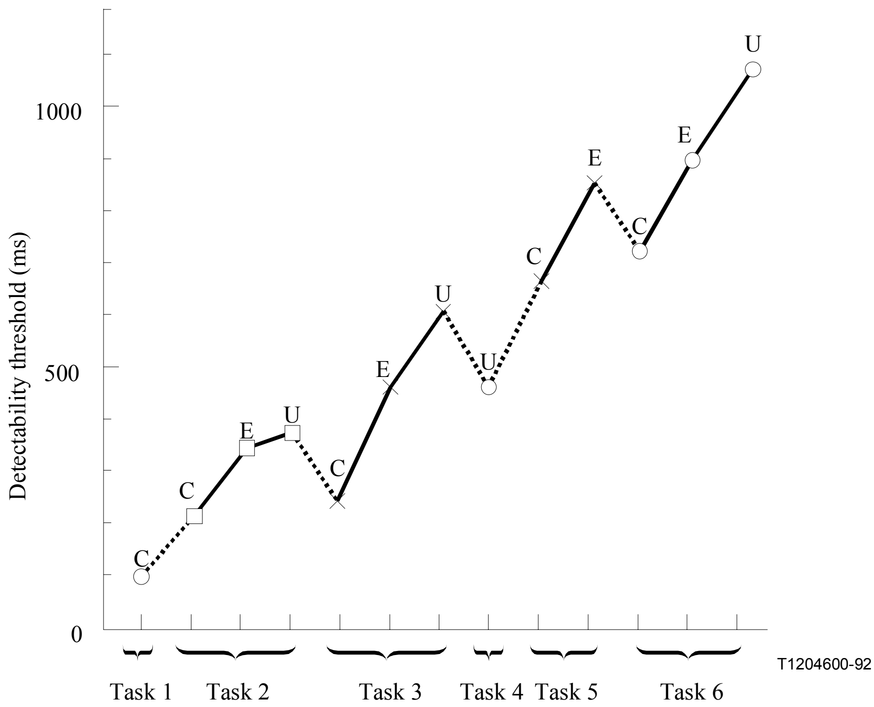
– Task 3: Complete words with lost letters as quickly as possible by exchanging information.

– Task 4: Verify city names as quickly as possible in turn.

– Task 5: Determine the shape of a figure by receiving oral information.

– Task 6: Free conversation.

Subjective opinion tests were performed and delay detectability thresholds. Mean Opinion Scores (MOS) and conversation efficiency were obtained. Figure B.4 shows detectability thresholds for various conversational tasks. The results show that the subjective quality as a function of delay varies depending on a conversational mode and subject group (trained, untrained).



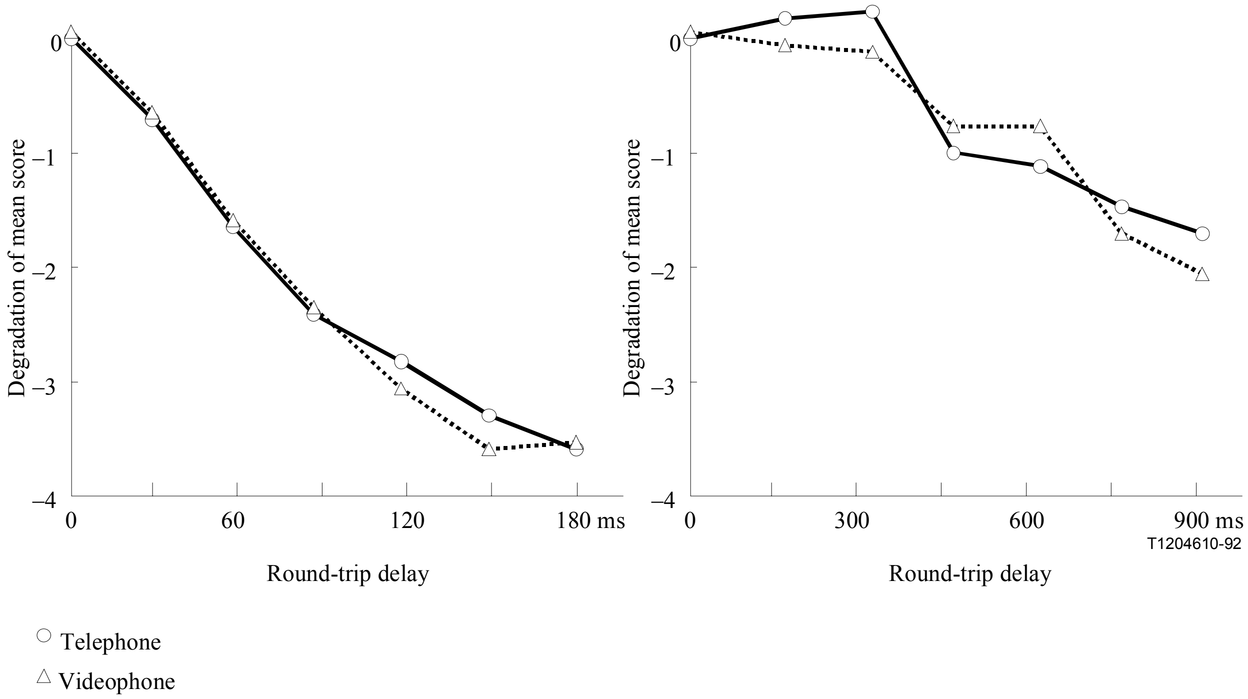
U Untrained (Businessmen, housewives, students)
C Trained (Crews)
E Untrained (Laboratory employees)

Figure B.4/G.114 – Detectability thresholds for various conversational models

In Figure B.4, the detectability threshold for round-trip dealy was defined as the delay detected by 50% of a task's subjects and provides some guidance to network planners in providing acceptable service to the user.

B.2.3 Interaction between delay and user applications

Tests were performed to assess the interaction between delay and user applications. In these tests a comparison of telephone conversations with videophone were made and it was shown that there is little difference between both types of connection. Figure B.5 shows the degradations of MOS, using a condition without delay as anchor.



a) Task 1 **b) Task 6**

Figure B.5/G.114 – Effects of delay on communications quality for telephone and videophone

A methodology for objective assessment of the effects of delay on speech communication in real networks was derived using the results of the above subjective experiments.

The information on temporal characteristics and their correlation to subjective opinions was extracted from the subjective data. This data was then used to formulate equations predicting detectability threshold and MOS as a function of delay. The effects of the delay on performance in commercial networks can be estimated by measuring the basic temporal parameters from the real life traffic and then using this data to calculate the objective measures applying the experimentally derived equations.

Table B.1 presents an example of the results obtained using this methodology for a commercial circuit.

In 1992, a study was performed to assess the subjective impact of end-to-end transmission delay in audiovisual communications. The experimental conditions included three point-to-point videophone connections with 200, 450 and 700 ms of one-way transmission delays. Subjects engaged in a series of five-minute long conversations and were interrogated at the end of each condition, as well as after the whole session. The results are summarized in Table B.2. Similar results were obtained from a videotelephony test.

Table B.1/G.114 – Effect of delay on speech quality in a real network

Conversation mode \ Quality		Cumulative distribution (%)	Detectability threshold (Round-trip dealy) (ms)
Type of commercial call	Task 1	0.1	90
	Task 2	1	210
	Task 3	9	290
	Task 4	21	480
	Task 5	86	680
	Task 6	80	740

Table B.2/G.114 – Variation of subjective performance for three end-to-end videophone connections

	One-way transmission delays		
	200 ms	**450 ms**	**700 ms**
MOS connection quality	3.74 ± 0.52	3.69 ± 0.51	3.48 ± 0.48
MOS ease of interruption	4.00 ± 0.55	3.79 ± 0.53	3.56 ± 0.49
Communication difficulty	28 ± 4%	35 ± 5%	46 ± 6%
Connection acceptability	80 ± 11%	78 ± 11%	73 ± 10%
NOTE – MOS values were derived on the basis of a five-point (1 to 5) scale. All errors are defined at a 95% level of confidence.			

B.3 Summary and conclusions

The transmission impairments associated with long delay circuits are best analysed by separating the echo-induced degradation and the subjective difficulty due to pure delay. Appropriate use of echo cancellers has been shown to indeed provide international or national satellite connections yielding quality and performance practically equivalent to the terrestrial connections for telephony. These results only refer to electric echo and additional studies are necessary to determine the effect of acoustic echo.

Thus, under these conditions, the dominant impairments are associated with the pure delay component.

Recently presented information suggests that:

– the effects of pure delay (no echo) on conversation dynamics can be detected well below 400 ms one-way delay if subjective experiments employ highly interactive tasks and subjective measures related to specific conversational difficulties, such as ability to interrupt, are used.

– the effects of pure delay (no echo) on speech quality appear to moderately increase as the delay is increased.

However, as a standard set of tests has not been agreed to, obtained experimental results depend upon the type of activity selected to evaluate the impact of delay and experimental results vary significantly from laboratory to laboratory. Thus, designers must determine the type of services, and hence the communication interactivity needs, that will be carried if the performance of the system is to be appropriately evaluated.

APPENDIX I

Delay introduced by coder-related processing

Table I.1/G.114 – Delay values for coders in wirebound applications

Coder type	Rate (kbit/s)	Frame size (ms)	Look-ahead (ms)	Mean one-way delay introduced by coder-related processing (ms)	Reference
PCM	64	0.125	0	0.375	G.711, G.712
ADPCM	40	0.125	0	0.375	G.726, G.727
ADPCM	32	0.125	0	0.375	G.721(1988), G.726, G.727
ADPCM	24	0.125	0	0.375	G.726, G.727
ADPCM	16	0.125	0	0.375	G.726, G.727
LD-CELP	16	0.625	0	1.875	G.728
LD-CELP	12.8	0.625	0	1.875	G.728
CS-ACELP	8	10	5	35	G.729
VSELP	7.95	20	0	60	IS-54-B, TIA
ACELP	7.4	20	5	65	IS-641, TIA
QCELP	8	20	0	60	IS-96-A
RCELP	8	20	10	70	IS-127
VSELP	6.7	20	5	65	Japanese PDC
RPE-LTP	13	20	0	60	GSM 06.10, Full-rate
VSELP	5.6	20	0	60	GSM 06.20, Half-rate
ACELP	12.2	20	0	60	GSM 06.60, Enhanced FR
ACELP	5.3	30	7.5	97.5	G.723.1
MP-MLQ	6.3	30	7.5	97.5	G.723.1

NOTE 1 – The PCM coder converts from analogue to digital and vice versa while all other coders refer to the PCM domain; for PCM in the analogue domain additional delay is incurred (0.375 ms).

NOTE 2 – For wirebound applications the mean one-way delay introduced by codec-related processing = 3 × frame size + look-ahead (see A.4.1).

Table I.2/G.114 – Delay values for coders in mobile or cordless applications

Coder type	Rate (kbit/s)	Frame size (ms)	Look-ahead (ms)	Air interface framing ms	Mean one-way delay introduced by coder-related processing (ms)	Reference
PCM	64	0.125	0	See Note 3		G.711, G.712
ADPCM	40	0.125	0	See Note 3		G.726, G.727
ADPCM	32	0.125	0	13.625	14	G.721(1988), G.726, G.727, DECT
ADPCM	24	0.125	0	See Note 3		G.726, G.727
ADPCM	16	0.125	0	See Note 3		G.726, G.727
LD-CELP	16	0.625	0	See Note 3		G.728
LD-CELP	12.8	0.625	0	See Note 3		G.728
CS-ACELP	8	10	5	See Note 3		G.729
VSELP	7.95	20	0			IS-54-B, TIA
ACELP	7.4	20	5			IS-641, TIA
QCELP	8	20	0			IS-96-A
RCELP	8	20	10			IS-127
VSELP	6.7	20	5			Japanese PDC
RPE-LTP	13	20	0	35	95	GSM 06.10, Full-rate
VSELP	5.6	20	0	35	95	GSM 06.20, Half-rate
ACELP	12.2	20	0	35	95	GSM 06.60, Enhanced FR
ACELP	5.3	30	7.5	See Note 3		G.723.1
MP-MLQ	6.3	30	7.5	See Note 3		G.723.1

NOTE 1 – The PCM coder converts from analogue to digital and vice versa while all other coders refer to the PCM domain; for PCM in the analogue domain, additional delay is incurred (0.375 ms).

NOTE 2 – For mobile or cordless applications the mean one-way delay introduced by codec-related processing = 3 × frame size + look-ahead + air interface framing (see A.4.2)

NOTE 3 – For the marked types of coders, Study Group 12 is not aware of any mobile or cordless application.

Table I.3/G.114 – Delay values for coders in IP-based applications (one frame per packet)

Coder type	Rate (kbit/s)	Frame size (ms)	Look-ahead (ms)	Mean one-way delay introduced by coder-related processing (ms) (see Note 2)		Reference
				Minimum	Maximum	
PCM	64	0.125	0	0.25	0.375	G.711, G.712
ADPCM	40	0.125	0	0.25	0.375	G.726, G.727
ADPCM	32	0.125	0	0.25	0.375	G.721(1988), G.726, G.727
ADPCM	24	0.125	0	0.25	0.375	G.726, G.727
ADPCM	16	0.125	0	0.25	0.375	G.726, G.727
LD-CELP	16	0.625	0	1.25	1.875	G.728
LD-CELP	12.8	0.625	0	1.25	1.875	G.728
CS-ACELP	8	10	5	25	35	G.729
VSELP	7.95	20	0	40	60	IS-54-B, TIA
ACELP	7.4	20	5	45	65	IS-641, TIA
QCELP	8	20	0	40	60	IS-96-A
RCELP	8	20	10	50	70	IS-127
VSELP	6.7	20	5	45	65	Japanese PDC
RPE-LTP	13	20	0	40	60	GSM 06.10, Full-rate
VSELP	5.6	20	0	40	60	GSM 06.20, Half-rate
ACELP	12.2	20	0	40	60	GSM 06.60, Enhanced FR
ACELP	5.3	30	7.5	67.5	97.5	G.723.1
MP-MLQ	6.3	30	7.5	67.5	97.5	G.723.1

NOTE 1 – The PCM codec converts from analogue to digital and vice versa while all other coders refer to the PCM domain; for PCM in the analogue domain additional delay is incurred (0.375 ms).

NOTE 2 – For IP-related applications the mean one-way delay introduced by codec-related processing:

= 2 × frame size + look-ahead (minimum, see A.4.3)
= 3 × frame size + look-ahead (maximum, see A.4.3).

Table I.4/G.114 – Delay values for coders in IP-based applications (multiple frames per packet)

Coder type	Rate (kbit/s)	Frame size (ms)	Look-ahead (ms)	Mean one-way delay introduced by coder-related processing (ms) (see Note 2)		Reference
				Minimum	Maximum	
PCM	64	0.125	0	(N + 1) × 0.125	(2N + 1) × 0.125	G.711, G.712
ADPCM	40	0.125	0	(N + 1) × 0.125	(2N + 1) × 0.125	G.726, G.727
ADPCM	32	0.125	0	(N + 1) × 0.125	(2N + 1) × 0.125	G.721(1988), G.726, G.727
ADPCM	24	0.125	0	(N + 1) × 0.125	(2N + 1) × 0.125	G.726, G.727
ADPCM	16	0.125	0	(N + 1) × 0.125	(2N + 1) × 0.125	G.726, G.727
LD-CELP	16	0.625	0	(N + 1) × 0.625	(2N + 1) × 0.625	G.728
LD-CELP	12.8	0.625	0	(N + 1) × 0.625	(2N + 1) × 0.625	G.728
CS-ACELP	8	10	5	(N + 1) × 10 + 5	(2N + 1) × 10 + 5	G.729
VSELP	7.95	20	0	(N + 1) × 20	(2N + 1) × 20	IS-54-B, TIA
ACELP	7.4	20	5	(N + 1) × 20 + 5	(2N + 1) × 20 + 5	IS-641, TIA
QCELP	8	20	0	(N + 1) × 20	(2N + 1) × 20	IS-96-A
RCELP	8	20	10	(N + 1) × 20 + 10	(2N + 1) × 20 + 10	IS-127
VSELP	6,7	20	5	(N + 1) × 20 + 5	(2N + 1) × 20 + 5	Japanese PDC
RPE-LTP	13	20	0	(N + 1) × 20	(2N + 1) × 20	GSM 06.10, Full-rate
VSELP	5.6	20	0	(N + 1) × 20	(2N+1) x 20	GSM 06.20, Half-rate
ACELP	12.2	20	0	(N + 1) × 20	(2N + 1) × 20	GSM 06.60, Enhanced FR
ACELP	5.3	30	7.5	(N + 1) × 30 + 7.5	(2N + 1) × 30 + 7.5	G.723.1
MP-MLQ	6.3	30	7.5	(N + 1) × 30 + 7.5	(2N + 1) × 30 + 7.5	G.723.1

NOTE 1 – The PCM codec converts from analogue to digital and vice versa while all other coders refer to the PCM domain; for PCM in the analogue domain additional delay is incurred (0.375 ms).

NOTE 2 – For IP-related applications with multiple frames per packet the mean one-way delay introduced by codec-related processing can be calculated as follows:

= (N + 1) × frame size + look-ahead (minimum, see A.4.4)
= (2N + 1) × frame size + look-ahead (maximum, see A.4.4).

NOTE 3 – N = number of frames per packet.

APPENDIX II

Bibliography

– CCITT Recommendation G.711 (1988), *Pulse code modulation (PCM) of voice frequencies*.

– CCITT Recommendation G.721 (1988) (withdrawn 1993), *32 kbit/s adaptive differential pulse code modulation (ADPCM)*.

– ITU-T Recommendation G.723.1 (1996), *Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s*.

– CCITT Recommendation G.726 (1990), *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)*.

– CCITT Recommendation G.727 (1990), *5-, 4-, 3- and 2-bits sample embedded adaptive differential pulse code modulation (ADPCM)*.

– CCITT Recommendation G.728 (1992), *Coding of speech at 16 kbit/s using low-delay code excited linear prediction.*

– ITU-T Recommendation G.728 Annex H (1999), *Variable bit rate LD-CELP operation mainly for DCME at rates less than 16 kbit/s*.

– ITU-T Recommendation G.729 (1996), *Coding of speech at 8 kbit/s using conjugate-structure algebraic-code-excited linear-prediction (CS-ACELP)*.

– ITU-T Recommendation G.729 Annex D (1998), *6.4 kbit/s CS-ACELP speech coding algorithm*.

– ITU-T Recommendation G.729 Annex E (1998), *11.8 kbit/s CS-ACELP speech coding algorithm*.

For the purposes of this ITU-T Recommendation, the following standards should be considered together as a package:

– TIA/EIA/IS-54-B (1992), *Cellular System Dual-Mode Mobile Station – Base Station Compatibility Standard* (upgraded to TIA/EIA-627 in June 1996).

– TIA/EIA-627 (1996), *800 MHz Cellular System, TDMA Radio Interface, Dual-Mode Mobile Station – Base Station Compatibility Standard*.

– ANSI/TIA/EIA-96-C (1998), *Speech Service Option Standard for Wideband Spread Spectrum Systems*.

– TIA/EIA/IS-127 (1997), *Enhanced Variable Rate Codec, Speech Service Option 3 for Wideband Spread Spectrum Digital Systems*.

– TIA/EIA/IS-641-A (1998), *TDMA Cellular/PCS-Radio Interface – Enhanced Full-Rate Speech Codec*.

– ETSI ETS 300 175-8 VI.4.2 (1999), *Digital Enhanced Cordless Telecommunications (DECT); Common Interface (CI); Part 8: Speech Coding and Transmission*.

For the purposes of this ITU-T Recommendation, the following standards should be considered together as a package:

– ETSI EN 300 961 V7.0.2 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Transcoding (GSM 06.10 version 7.0.2 Release 1998)*.

– ETSI EN 300 962 V7.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Substitution and muting of lost frames for full rate speech channels* (GSM 06.11 version 7.0.1 Release 1998).

– ETSI EN 300 963 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Comfort noise aspect for full rate speech traffic channels* (GSM 06.12 version 6.0.1 Release 1997).

– ETSI EN 300 964 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Discontinuous Transmission (DTX) for full rate speech traffic channels* (GSM 06.31 version 6.0.1 Release 1997).

– ETSI EN 300 965 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Full rate speech; Voice Activity Detector (VAD) for full rate speech traffic channels* (GSM 06.32 version 6.0.1 Release 1997).

For the purposes of this ITU-T Recommendation the following standards should be considered together as a package:

– ETSI EN 300 969 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Half rate speech transcoding* (GSM 06.20 version 6.0.1 Release 1997).

– ETSI EN 300 970 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Substitution and muting of lost frames for half rate speech traffic channels (GSM 06.21 version 6.0.1 Release 1997)*.

– ETSI EN 300 971 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Comfort noise aspects for the half rate speech traffic channels* (GSM 06.22 version 6.0.1 Release 1997).

– ETSI EN 300 972 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Discontinuous Transmission (DTX) for half rate speech traffic channels* (GSM 06.41 version 6.0.1 Release 1997).

– ETSI EN 300 973 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Half rate speech; Voice Activity Detector (VAD) for half rate speech traffic channels* (GSM 06.42 version 6.0.1 Release 1997).

For the purposes of this ITU-T Recommendation the following standards should be considered together as a package:

– ETSI EN 300 726 V7.0.2 (1999), *Digital cellular telecommunications system (Phase 2+); Enhanced Full Rate (EFR) speech transcoding* (GSM 06.60 version 7.0.2 Release 1998).

– ETSI EN 300 727 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Substitution and muting of lost frames for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.61 version 6.0.1 Release 1997).

– ETSI EN 300 728 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Comfort noise aspects for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.62 version 6.0.1 Release 1997).

– ETSI EN 300 729 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Discontinuous Transmission (DTX) for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.81 version 6.0.1 Release 1997).

– ETSI EN 300 730 V6.0.1 (1999), *Digital cellular telecommunications system (Phase 2+); Voice Activity Detector (VAD) for Enhanced Full Rate (EFR) speech traffic channels* (GSM 06.82 version 6.0.1 Release 1997).

– ARIB: RCR STD-27 H, Fascicle 1 (February 2, 1999), *Personal Digital Cellular Telecommunication System ARIB Standard*.

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