ATIS-0100501.1994(R2013)

Tandem Encoding Limits for 32-kbit/s ADPCM

AMERICAN NATIONAL STANDARD FOR TELECOMMUNICATIONS

As a leading technology and solutions development organization, ATIS brings together the top global ICT companies to advance the industry's most-pressing business priorities. Through ATIS committees and forums, nearly 200 companies address cloud services, device solutions, emergency services, M2M communications, cyber security, ehealth, network evolution, quality of service, billing support, operations, and more. These priorities follow a fast-track development lifecycle – from design and innovation through solutions that include standards, specifications, requirements, business use cases, software toolkits, and interoperability testing.

ATIS is accredited by the American National Standards Institute (ANSI). ATIS is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a founding Partner of oneM2M, a member and major U.S. contributor to the International Telecommunication Union (ITU) Radio and Telecommunications sectors, and a member of the Inter-American Telecommunication Commission (CITEL). For more information, visit < www.atis.org >.

AMERICAN NATIONAL STANDARD

Approval of an American National Standard requires review by ANSI that the requirements for due process, consensus, and other criteria for approval have been met by the standards developer.

Consensus is established when, in the judgment of the ANSI Board of Standards Review, substantial agreement has been reached by directly and materially affected interests. Substantial agreement means much more than a simple majority, but not necessarily unanimity. Consensus requires that all views and objections be considered, and that a concerted effort be made towards their resolution.

The use of American National Standards is completely voluntary; their existence does not in any respect preclude anyone, whether he has approved the standards or not, from manufacturing, marketing, purchasing, or using products, processes, or procedures not conforming to the standards.

The American National Standards Institute does not develop standards and will in no circumstances give an interpretation of any American National Standard. Moreover, no person shall have the right or authority to issue an interpretation of an American National Standard in the name of the American National Standards Institute. Requests for interpretations should be addressed to the secretariat or sponsor whose name appears on the title page of this standard.

CAUTION NOTICE: This American National Standard may be revised or withdrawn at any time. The procedures of the American National Standards Institute require that action be taken periodically to reaffirm, revise, or withdraw this standard. Purchasers of American National Standards may receive current information on all standards by calling or writing the American National Standards Institute.

Notice of Disclaimer & Limitation of Liability

The information provided in this document is directed solely to professionals who have the appropriate degree of experience to understand and interpret its contents in accordance with generally accepted engineering or other professional standards and applicable regulations. No recommendation as to products or vendors is made or should be implied.

NO REPRESENTATION OR WARRANTY IS MADE THAT THE INFORMATION IS TECHNICALLY ACCURATE OR SUFFICIENT OR CONFORMS TO ANY STATUTE, GOVERNMENTAL RULE OR REGULATION, AND FURTHER, NO REPRESENTATION OR WARRANTY IS MADE OFMERCHANTABILITY OR FITNESS FOR ANY PARTICULAR PURPOSE OR AGAINST INFRINGEMENT OF INTELLECTUAL PROPERTY RIGHTS. ATIS SHALL NOT BE LIABLE, BEYOND THE AMOUNT OF ANY SUM RECEIVED IN PAYMENT BY ATIS FOR THIS DOCUMENT, AND IN NO EVENT SHALL ATIS BE LIABLE FOR LOST PROFITS OR OTHER INCIDENTAL OR CONSEQUENTIAL DAMAGES. ATIS EXPRESSLY ADVISES THAT ANY AND ALL USE OF OR RELIANCE UPON THE INFORMATION PROVIDED IN THIS DOCUMENT IS AT THE RISK OF THE USER.

NOTE - The user's attention is called to the possibility that compliance with this standard may require use of an invention covered by patent rights. By publication of this standard, no position is taken with respect to whether use of an invention covered by patent rights will be required, and if any such use is required no position is taken regarding the validity of this claim or any patent rights in connection therewith. Please refer to [http://www.atis.org/legal/patentinfo.asp] to determine if any statement has been filed by a patent holder indicating a willingness to grant a license either without compensation or on reasonable and non-discriminatory terms and conditions to applicants desiring to obtain a license.

ATIS-0100501.1994(R2013), *Tandem Encoding Limits for 32 kbit/s ADPCM*

Is an American National Standard developed by the **ATIS Network Performance, Reliability and Quality of Service Committee (PRQC)**.

Published by

Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

ANSI
ATIS-0100501.1994 (R2013)

(Formerly T1.501-1994, Revision of T1.501-1988)

American National Standard
for Telecommunications –

Network Performance – Tandem Encoding Limits for 32-kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM)

Secretariat

Alliance for Telecommunications Industry Solutions

Approved October 10, 1994

American National Standards Institute, Inc.

Abstract

This standard specifies the limitations on the maximum number of ITU-T Recommendation G.726 32-kbit/s adaptive differential pulse-code modulation (ADPCM) encodings allowable in 4-kHz voice grade network connections. This allows for the realization of the possible economic advantages of ADPCM use while retaining quality transmission performance capability. The applicable limits are specified as a function of the category of signal transported. Performance impact information and functional usage guidelines are provided to give guidance on achieving compliance with the limits in the standard, but these are not a part of the standard.

Table of Contents

Foreword (This foreword is not part of American National Standard T1.501-1994.)

The specification of 32-kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM) tandem encoding limits was initiated under the auspices of the Accredited Standards Committee on Telecommunications, T1, which is sponsored by the Alliance for Telecommunications Industry Solutions (ATIS). ADPCM is one of a group of emerging technologies, each of which provides the capability for more efficient use of digital facility capacity through the use of processing techniques. There is an industry desire to take advantage of the potential benefits of the introduction of new technology without negatively affecting performance. This standard examines performance relative to the 32-kbit/s ADPCM algorithm as specified in ITU-T Recommendation G.726, *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM*). It is intended to provide the orderly management of the introduction of this technology so as to meet the performance needs of end-users.

Traditional analog transmission parameters do not characterize adequately the effect of ADPCM on performance. As a result, performance is assured by establishing a limit on the number of tandem encodings for two fundamental categories of signals that encompass the signal types transported by both public- and private-network connections. The use of this standard will help ensure that the performance as seen by the end-user will be satisfactory. This standard allows for the realization of the possible economic advantages of ADPCM use while retaining quality transmission performance capability.

The specifications in this standard apply only to 32-kbit/s ADPCM algorithms defined in ITU-T Recommendation G.726. The performance impact of other ADPCM algorithms and other compression methods are currently under study by Working Group T1A1.7 and will be addressed in a future technical report.

This standard contains three annexes. Annexes A – C are informative and are not considered a part of this standard.

Suggestions for improvement of this standard are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions (ATIS), Committee T1 Secretariat, 1200 G Street, NW, Suite 500, Washington, DC 20005.

This standard was processed and approved for submittal to ANSI by Accredited Standards Committee on Telecommunications, T1. Committee approval of this standard does not necessarily imply that all committee members voted for its approval. At the time it approved this standard, Committee T1 had the following members:

Arthur K. Reilly, Chair
Gerald H. Peterson, Vice-Chair
O. J. Gusella, Jr., Secretary

Barry Lerich, Senior Editor

Organization Represented *Name of Representative*

EXCHANGE CARRIERS

Ameritech Services, Inc.	Laurence A. Young
	Stephen P. Murphy (Alt.)
Bell Atlantic	John W. Seazholtz
	Roger Nucho (Alt.)
Bellcore	James C. Staats
	E. R. Hapeman (Alt.)
BellSouth Telecommunications, Inc.	William J. McNamara, III
	Malcolm Threlkeld, Jr. (Alt.)
Cincinnati Bell Telephone	Thomas C. Grimes
	Renee W. Cagle (Alt.)
GTE Telephone Operations	Bernard J. Harris
	Richard L. Cochran (Alt.)
National Telephone Cooperative Association	Joseph M. Flanigan
NYNEX	James F. Baskin
	Jim Papadopoulos (Alt.)
Pacific Bell	Sal R. Tesoro
Puerto Rico Telephone Company	Segundo Ruiz
	Alberto E. Morales (Alt.)
Southwestern Bell Corporation	C. C. Bailey
	Joseph Mendoza (Alt.)
Sprint – Local Telecommunications Division	Robert P. McCabe
	Harold L. Fuller (Alt.)
US Telephone Association (USTA)	Dennis Byrne
	Paul Hart (Alt.)
US WEST	James L. Eitel
	Darryl Debault (Alt.)

INTEREXCHANGE CARRIERS

AT&T Communications	Charles A. Dvorak
	Dennis Thovson (Alt.)
Comsat Corporation	Mark T. Neibert
	Thanos Kipreos (Alt.)
MCI Telecommunications Corporation	Jim Joerger
	Peter Guggina (Alt.)
Sprint – Long Distance Division	Tom G. Croda
	Peter J. May (Alt.)
Stentor Resource Centre, Inc.	Michel Duchesne
	B. Sambasivan (Alt.)
Unitel Communications, Inc.	David H. Whyte
	George Tadros (Alt.)
Wiltel, Inc.	Robert Bentley
	Howard Meiseles (Alt.)

MANUFACTURERS

ADC Telecommunications, Inc.	Ron Weitnauer
	Don Berryman (Alt.)
Alcatel Network Systems (ANS)	Robert Cubbage
	Dale Krisher (Alt.)
AMP, Inc.	George Lawrence
	Jack Bradbery (Alt.)
Apple Computer, Inc.	David Michael
	Karen Higginbottom (Alt.)
Ascom Timeplex, Inc.	L. H. Eberl
	Richard Koepper (Alt.)
AT&T Network Systems	John H. Bobsin
	Dave R. Andersen (Alt.)
DSC Communications Corporation	Peter Waal
	Allen Adams (Alt.)
ECI Telecom, Inc.	Ron Murphy
	Charles T. Throop (Alt.)
Ericsson, Inc.	Linda Troy
	Al Way (Alt.)
Fujitsu America, Inc.	Kenneth T. Coit
	Rodney Boehm (Alt.)
General DataComm, Inc.	Frederick Cronin
	Frederick Lucas (Alt.)
Harris Corporation	Allen Jackson
	Yogi Mistry (Alt.)

Hekimian Laboratories	William H. Duncan
	Mike F. Toohig (Alt.)
Hewlett-Packard	Don C. Loughry
	Richard van Gelder (Alt.)
IBM Corporation	Neville L. Golding
	William Bergman (Alt.)
Mitel Corporation	John Needham
	F. Audet (Alt.)
Motorola, Inc.	David Morgan
	Gail Smith (Alt.)
NEC America, Inc.	Art Graham
	Donovan Nak (Alt.)
Northern Telecom, Inc.	Mel N. Woinsky
	Subhash Patel (Alt.)
Novatel Communications, Ltd.	Allan Angus
Omnipoint Corporation	Gary K. Jones
	Logan Scott (Alt.)
Picturetel Corporation	Marshall Schachtman
	Antony Crossman (Alt.)
Racal-Datacom	Paul Nikolich
Reliance Comm/Tec	Mark Scott
	Leroy Baker (Alt.)
Rockwell International	Quent C. Cassen
	Carl J. Stehman (Alt.)
Siemens Stromberg-Carlson	Michael A. Pierce
	Robert Poignant (Alt.)
Telecom Solutions	M. J. Narasimha
	Don Chislow (Alt.)
Telecommunications Techniques	Bernard E. Worne
Teleos Communications, Inc.	Hascall Sharp
	Ken Araujo (Alt.)
Tellabs Operations, Inc.	Charles Rohrs
	Michael J. Birck (Alt.)
Transwitch Corporation	Daniel C. Upp
	Praveen Goli (Alt.)

GENERAL INTEREST

Ashford Associates	Donald A. Ashford
Brooktree Associates	Douglas M. Brady
	Tim Lindenfelser (Alt.)
BT North America, Inc.	Ron Stotz
C.S.I. Telecommunications	Michael S. Newman
	William J. Buckley (Alt.)
Cable Television Labs, Inc.	Rhonda Hilton
	James S. Meditch (Alt.)
Capital Cities/ABC, Inc.	Warner W. Johnston
Defense Information Systems Agency	C. Joseph Pasquariello
	Gary L. Koerner (Alt.)
EDS Corporation	Douglas Zolnick
GTE Mobile Communications	John C. Chiang
	Steve Pankow (Alt.)
International Communications Association	Edward F. Bonkowski
	Robert M. Eilers (Alt.)
National Communications System	Dennis Bodson
National Institute of Standards and Technology	David Cypher
	Leslie A. Collica (Alt.)
National Telecommunications and Information Administration/Institute for Telecommunication Sciences (NTIA/ITS)	William F. Utlaut
	Neal B. Seitz (Alt.)
NTT America, Inc.	Hideo Yamamoto
	Naobumi Kanemaki (Alt.)
Rural Electrification Administration	Donald M. Van Bellinger
	George J. Bagnall (Alt.)
U. S. General Services Administration	Douglas K. Arai
	Patrick Plunkett (Alt.)

At the time it approved this standard, Technical Subcommittee for Performance and Signal Processing, T1A1, had the following officers and members:

Charles A. Dvorak, Chairman
Joe A. Zebarth, Secretary

Organization Represented	*Name of Representative*
Alcatel Network Systems (ANS)	Kevin Pickles
	Albert Azzam (Alt.)
Ameritech Services, Inc.	Michael S. Peats
	Lorence F. Brown (Alt.)
AT&T Communications	Anthony Schiano
	Charles A. Dvorak (Alt.)
AT&T Network Systems	Robert B. Waller
	Carl R. Posthuma (Alt.)
Bell Atlantic	Max Roesch
	Lita B. Gwinn (Alt.)
Bellcore	Ralph E. Jensen
	W. Garry Couch (Alt.)
BellSouth Telecommunications, Inc.	Hal B. Holton
	Gregory A. Wos (Alt.)
Brooktree Corporation	Douglas M. Brady
C.S.I. Telecommunications	Michael S. Newman
Capital Cities/ABC, Inc.	Warner W. Johnston
Compression Labs, Inc.	Dan Klenke
	Padmanabha Rao (Alt.)
Comsat Corporation	Thanos Kipreos
	Dave Weinreich (Alt.)
Defense Information Systems Agency	Gary L. Koerner
Ericsson, Inc.	Sanjay Dhawan
	Sanjay Bhat (Alt.)
Fujitsu America, Inc.	Tetsuo Soejima
	Kanji Hayashi (Alt.)
General DataComm, Inc.	David Moon
	Frederick Lucas (Alt.)
GTE Telephone Operations	Richard L. Cochran
	Rich G. Quirk (Alt.)
IBM Corporation	Frank R. Koperda
	Allen L. Roginsky (Alt.)
MCI Telecommunications Corporation	Michael Varrassi
	Morton Blanchard (Alt.)
National Communications System	Gary Rekstad
	Granger Kelley (Alt.)
National Telecommunications and Information Administration/Institute for Telecommunication Sciences (NTIA/ITS)	Neal B. Seitz
	Randall S. Bloomfield (Alt.)
NEC America, Inc.	Donovan Nak
Newbridge Networks Corporation	Zlatko Krstulich
	Francois Bessette (Alt.)
Northern Telecom, Inc.	Mel N. Woinsky
	Andy Niedzwiecki (Alt.)
NYNEX	F. T. Burns
	John McDonough (Alt.)
Pacific Bell	Ali Zolfaghari
	Sal R. Tesoro (Alt.)
Picturetel Corporation	Antony Crossman
	Marshall Schachtman (Alt.)
Rockwell International	Tom Geary
	Glen R. Griffith (Alt.)
Siemens Stromberg-Carlson	Gil Hassell
	Michael A. Pierce (Alt.)
Southwestern Bell Corporation	John E. Roquet
	Robert J. Hall (Alt.)

Sprint – Local Telecommunications Division	Tim Bower
	Jim McNaughton (Alt.)
Sprint – Long Distance Division	Cannon Hwu
	James Lord (Alt.)
Stentor Resource Centre, Inc.	J. A. Zebarth
	B. M. Murphy (Alt.)
Telecom Solutions	M. J. Narasimha
	Kishan Shenoi (Alt.)
Telecommunications Techniques	Bernard E. Worne
Teleos Communications, Inc.	Hascall Sharp
	Ken Araujo (Alt.)
Tellabs Operations, Inc.	Maurice Givens
	Jim Mills (Alt.)
3M	Peter K. Cencer
	Bruce R. Norton (Alt.)
Unitel Communications, Inc.	David H. Whyte
US Telephone Association (USTA)	Robert Creighton
US West	Kipling E. Ferguson
	Randon K. Sheridan (Alt.)
Wiltel, Inc.	Howard Meiseles
	Mark Elden (Alt.)

Working Group T1A1.1, Voice and Voiceband Data, which developed this standard, had the following officers and participants:

Peter Cencer, Chairman
Michael Kalb, Vice-Chairman
Joe Zebarth, Secretary

AMERICAN NATIONAL STANDARD **ATIS-0100501.1994** (R2013)

American National Standard
for Telecommunications –

Network Performance – Tandem Encoding Limits for 32-kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM)

1 Scope, purpose, and application

1.1 Scope

This standard addresses the use of 32-kbit/s Adaptive Differential Pulse-Code Modulation (G.726 32-kbit/s ADPCM)[1)] in 4-kHz voiceband network connections and limits the number of G.726 32-kbit/s ADPCM links allowed. It relates to the transport of signals processed by the 32-kbit/s ADPCM algorithm specified in ITU-T Recommendation G.726. Although G.726 defines other ADPCM algorithms designed for operation at 16, 24, and 40 kbit/s, the performance implications of these algorithms are not addressed in this standard. This standard does not purport to describe the service offerings of any particular carrier.

1.2 Purpose

This standard stipulates appropriate limits on the use of G.726 32-kbit/s ADPCM while retaining, for all carriers, the capability to provide quality transmission performance to end-users.

1.3 Application

The standard applies to end-to-end voiceband connections established on either public switched telecommunications networks (PSTNs) or private networks. The tandem encoding limits of this standard apply to exchange carriers (ECs), interexchange carriers (ICs) and private network providers.

2 Normative references

The following standards and publications contain provisions which, through reference in this text, constitute provisions of this American National Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this American National Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

ITU-T Recommendation G.113, *Transmission impairments*. Blue Book, Volume III – Fascicle III.1, Melbourne, November 1988[2)]

ITU-T Recommendation G.726, *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM).* Geneva, 1990[2)]

ITU-T Recommendation G.727, *5-, 4-, 3- and 2 bits embedded adaptive differential pulse code modulation (ADPCM).* Geneva, 1990[2)]

3 Definitions and acronyms

3.1 Definitions

The following definitions apply in this standard:

3.1.1 adaptive differential pulse-code modulation (ADPCM): A digital transcoding technique in which the difference between the actual and estimated value of the sample is encoded and transmitted to the decoder, where it is reconstructed into a PCM signal. In addition, it employs an adaptive quantizer and an adaptive predictor to follow changes in signal power and to respond to changes in the short-term spectrum of analog signals.

3.1.2 ADPCM link: Any transmission system, or part of a transmission system, that performs one unique PCM-ADPCM transcoding and a corresponding ADPCM-PCM transcoding.

3.1.3 bit error ratio (BER): The ratio of the number of bit errors to the total number of bits transmitted in a given time interval.

3.1.4 block error ratio (BLER): The ratio of the number of blocks that contain at least one bit in error to the total number of blocks transmitted in a given time interval.

3.1.5 network interface (NI): The point of demarcation between the carrier's facilities and the customer installation, which establishes the technical interface and division of operational responsibility. In this definition, as it applies in this document, the term "customer" refers to the end-user.

3.1.6 point of termination (POT): The point of demarcation between carriers, which establishes the technical interface and division of operational responsibility.

3.1.7 speech-to-speech-correlated-noise ratio (Q): The ratio of the speech power to the power of speech-correlated noise in a digital transmission system.

3.2 Acronyms

ADPCM	adaptive differential pulse-code modulation
ATIS	Alliance for Telecommunications Industry Solutions
BER	bit error ratio
BLER	block error ratio
CCITT	International Telegraph and Telephone Consultative Committee
DTMF	dual tone multi frequency
EC	exchange carrier
EO	end office
FSK	frequency shift keying
IC	interexchange carrier
ITU-T	International Telecommunications Union – Telecommunications Standardization Sector
NI	network interface
PCM	pulse-code modulation
PSTN	public switched telecommunications network
codec	coder-decoder
dB	decibel
kbit/s	kilobit per second
kHz	kiloHertz

ms millisecond

qdu quantizing distortion unit

4 Specification of limits

4.1 General

Traditional analog transmission parameters used in characterizing the performance of the various segments of the end-to-end connections are not adequate to retain quality transmission performance if G.726 32-kbit/s ADPCM equipment is deployed freely in the various segments. Currently, no standard test procedure is available to quantify the effect of G.726 32-kbit/s ADPCM links in a connection or part thereof.[3)] Therefore, the only method available to control the transmission performance impact of G.726 32-kbit/s ADPCM is to limit the number of G.726 32-kbit/s ADPCM links in connections. Information regarding the technical basis for the limits specified in this standard is included in annex A, ADPCM performance effects.

4.2 Categories of signals

The limitation on G.726 32-kbit/s ADPCM links in public or private networks depends upon the signal type being transmitted over network connections. Annex A provides information describing the effects of G.726 32-kbit/s ADPCM transcoding on several common signal types. The various signal types conveyed by voiceband networks fit into one of the following two categories:

Category 1: Voiceband (4 kHz) signals that can be transmitted with satisfactory performance over connections containing G.726 32-kbit/s ADPCM links;

Category 2: Voiceband (4 kHz) signals that cannot be transmitted with satisfactory performance over connections containing G.726 32-kbit/s ADPCM links.

These two categories apply to both PSTN and private-network connections. Criteria for satisfactory performance are application-dependent and ultimately determined by the end-user. In deriving the G.726 32-kbit/s ADPCM tandem encoding limits set forth in this standard, the following criteria were used:

Voice: The criterion is based on a measure called Q that captures the effect of quantizing noise on quality (see J. R. Cavanaugh in the bibliography). The criterion of Q ≥ 20 dB was used;

Voiceband data: Two criteria were used. Block error ratio (BLER) ≤ 10^{-2} (with a block size of 10^3 bits) and bit error ratio (BER) ≤ 10^{-5}. Whether BLER or BER is more meaningful depends upon the application. For transfers of large amounts of data with retransmittal protocols, BLER is appropriate. For facsimile applications and asynchronous character transfer, BER is more meaningful. The BER is usually the more stringent criterion;

Dual-tone multi frequnecy (DTMF) signals: The criterion of digit error ratio < 10^{-3} was used.

4.3 Tandem connecting arrangements

Two ADPCM links in tandem are said to be synchronously connected if the connecting transmission system is the equivalent of a 64-kbit/s PCM bit stream. Likewise, two ADPCM links are said to be asynchronously connected if they are connected by any other type of transmission system; in general, one which converts the signal-to-analog format at some point. The performance of two or more synchronously connected G.726 32-kbit/s ADPCM links is equivalent (except for delay) to one link if (1) the transmission of the G.726 32-kbit/s ADPCM links and the intermediate 64-kbit/s PCM links is error free[4)] and (2) the G.726 32-kbit/s ADPCM and intermediate 64-kbit/s PCM bit streams are not disturbed by digital signal processing. Any tandem connecting arrangement that employs PCM-modifying digital signal processing on the connecting link(s), such as that performed by digital echo cancelers or digital loss pads, will not satisfy the above constraints. If two G.726 32-kbit/s ADPCM links are connected by a 64-kbit/s transmission system that does not preserve bit integrity (i.e., it uses digital signal processing or bit-robbing, etc.) the net effect on performance will be bounded by the performance of one G.726 32-kbit/s ADPCM link and the performance of two asynchronously connected G.726 32-kbit/s ADPCM links. This effect, however, has not been well quantified; it is a subject for further study. Therefore, for purposes of this standard, this arrangement should be counted as equivalent to two asynchronously connected G.726 32-kbit/s ADPCM links.

4.4 Tandem encoding limits

The total number of G.726 32-kbit/s ADPCM links in connections shall be limited in order to provide the capability of quality transmission performance between end-users. These limits apply to PSTNs, private networks, and combinations thereof. The total number of asynchronously connected G.726 32-kbit/s ADPCM links in these network connections shall be limited as follows:

Category 1 signals: The limit is three asynchronously connected G.726 32-kbit/s ADPCM links;

Category 2 signals: The limit is zero G.726 32-kbit/s ADPCM links.

In some private network applications, it may be possible to customize the network to meet the end-user's specific application needs. In these cases, greater flexibility is possible, in terms of the number of G.726 32-kbit/s ADPCM links in tandem, while still satisfying the end-user's needs. An example of guidelines that may be used in such cases is contained in annex A.

4.5 Allocation of end-to-end limits

The allocation of the end-to-end limits presented in this standard is based on the assumption that G.726 32-kbit/s ADPCM is not deployed on the end-user's side of the network interface (NI). If G.726 32-kbit/s ADPCM is deployed on the end-user's side of the NI, the overall limits should be observed in order to provide satisfactory performance.

4.5.1 National connections

When a single carrier provides an end-to-end PSTN or private-network connection, the carrier shall implement the limits in clause 4.4. For connections with both EC- and IC-provided portions, the Category 1 signal limit of three is allocated, one each for the EC, IC and EC portions.

Each of the EC portions includes all components from the NI to the IC point of termination (POT). The Category 2 signal limit of zero requires that each provider's portion has no G.726 32-kbit/s ADPCM links. See annex B, Functional usage guidelines, for implementation information.

For both private and public switched connections, if an EC provides a G.726 32-kbit/s ADPCM-encoded signal at the EC-IC interface as a service option at the request of the IC, it will be considered as the IC's G.726 32-kbit/s ADPCM link.

4.5.2 International connections

Typical international connections would contain a combined EC and IC network in the national extension (NI to gateway). Based on the implementation of this standard, there would be zero, one, or two asynchronously connected G.726 32-kbit/s ADPCM links in the national extension. The allocation specified in the provisional planning rule, as described in ITU-T Recommendation G.113, allows at most one G.726 32-kbit/s ADPCM link in the national extension. However, an exception is noted that would allow at most two G.726 32-kbit/s ADPCM links with a clear indication that this is undesirable. Although the allocation of this standard allows up to a maximum of two links on the national extension of international connections, the frequency of occurrence of this maximum is expected to be low.

Annex A

(informative)

ADPCM performance effects

A.1 General

The impact of G.726 32-kbit/s ADPCM on the performance of a voiceband connection depends on the signal types transmitted (Category 1 or Category 2), the level and distribution of other analog impairments, the specific design of the terminal equipment and the number of asynchronously connected G.726 32-kbit/s ADPCM links in the connection. This annex is provided as an aid to the reader in understanding the limits in the standard. The information in this annex was derived from evaluations of specific modems and DTMF receivers of several manufacturers and may not describe performance of all available similar hardware.

A.2 References

The following documents are referenced in this annex:

[1] ANSI T1.508-1992, *Telecommunications – Network performance – Loss plan for evolving digital networks*

[2] ITU-T Recommendation G.113, *Transmission requirements*. Blue Book, Volume III – Fascicle III.1, Melbourne, 1988[2)]

[3] ITU-T Recommendation G.711, *Pulse code modulation (PCM) of voice frequencies*. Blue Book, Volume III – Fascicle III.4, Melbourne, 1988[2)]

[4] ITU-T Recommendation G.721, *32 kbit/s adaptive differential pulse code modulation (ADPCM).* Red Book, Volume III – Fascicle III.3, Malaga-Torremolinos, 1984[2)]

[5] ITU-T Recommendation G.721, *32 kbit/s adaptive differential pulse code modulation (ADPCM) (revised).* Blue Book, Volume III – Fascicle III.3, Melbourne, 1988[2)]

[6] ITU-T Recommendation V.17, *A 2-wire modem for facsimile applications with rates up to 14400 bits per second*. Geneva, 1991[2)]

[7] ITU-T Recommendation V.29*, 9600 bits per second modem standardized for use on point-to-point 4-wire leased telephone-type circuits*. Blue Book, Volume VIII – Fascicle VIII.1, Melbourne, 1988[2)]

[8] ITU-T Recommendation V.32*, A family of 2-wire, duplex modems operating at data signalling rates of up to 9600 bit/s for use on the general switched telephone network and on leased telephone-type circuits.* Blue Book, Volume VIII – Fascicle VIII.1, Melbourne, 1988[2)]

[9] ITU-T Recommendation V.32bis, *A duplex modem operating at data signalling rates of up to 14400 bits/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone type circuits*. Geneva, 1991[2)]

[10] ITU-T Draft Recommendation V.34, *A modem operating at data signalling rates of up to 28800 bits/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits.* 1994[2)]

[11] Carey, M. B.; Chen, H-T; Descloux, A.; Ingle, J. F.; Park, K.I., 1982/83 end office connection study; analog voice and voiceband data characterization of the public-switched network. *AT&T Bell Laboratories Technical Journal* 63, November 1984[5)]

[12] Daumer, W. R.; Sparrell, D*., FSK Enhancements for the proposed T1Y1 ADPCM algorithm.* Document T1Y1.2/86-006, January 1986[6)]

[13] Dimolitsas, S; Corcoran, F. L.; Onufry, M.; Suyderhoud, H. G., *Evaluation of ADPCM coders for digital circuit multiplication equipment.* COMSAT Technical Review, pp. 323 – 345, Fall 1987[7)]

[14] Kalb, M., ADPCM performance impact on voiceband data. In: *IEEE GLOBECOM 1985 Conference Record*, New York, IEEE, December 1985, pp. 1133-1137[8)]

[15] SG XVIII – Report R26, Source: CCITT Secretariat, *Report of the Meeting of Working Party XVIII/8 (Speech Processing).* July 1986[2)]

A.3 Definitions

A.3.1 quantizing distortion: The distortion introduced when an analog signal is encoded to digital format, then decoded to analog format. The quantizing distortion is the difference between the original analog signal and the analog signal resulting from the decoding process.

A.3.2 quantizing distortion unit (qdu): The distortion that is subjectively equivalent to the quantizing distortion introduced by an average 8-bit codec pair (A/D and D/A conversion; A-law or u-law) that complies with ITU-T Recommendation G.711 [3].[9)]

A.4 End-to-end performance criteria

In determining the tandem encoding limits, it was first necessary to establish a set of criteria for judging the acceptability of end-to-end performance. The voice criterion is based on a measure called Q. Various standard ITU-T recommendations have assigned values of Q to different processing techniques, including ADPCM, and have determined laws of addition for tandem devices and for combining with idle circuit noise. The ITU-T studies are based on the 32-kbit/s ADPCM algorithm defined in the 1984 ITU-T Recommendation G.721 [4] rather than that defined in the 1988 ITU-T Recommendation G.721 [5]. However, it has been shown that, except for the improved performance with respect to frequency-shift keying modems, the performance of the 1988 algorithm is essentially identical to that of the 1984 algorithm [12, 15]. The ITU-T, in Recommendation G.113 [2], has established a provisional end-to-end planning value of 14 quantizing distortion units (qdu), which corresponds to Q = 20 dB. A level of Q = 20 dB corresponds to the quantizing distortion (not idle-circuit noise) of 14 asynchronously tandemmed PCM (A/D, D/A) pairs. Studies have shown that in a typical network connection, a Q of 20 dB corresponds to a modest degradation in voice transmission quality. However, as Q decreases below 20 dB, there is a rapid and marked decrease in quality. On this basis, Q = 20 dB is a reasonable planning limit and is the voice criterion for this standard.

Delay also has an effect on overall transmission performance. Consider an end-to-end connection with up to three G.726 32-kbit/s ADPCM links in tandem. The additional round-trip delay consists of up to six G.726 32-kbit/s ADPCM encoding–decoding stages. As a general guide, if there are no active echo control devices in the connection, the additional round-trip delay should be less than 5 ms [1].

For voiceband data, two criteria are used: block error ratio (BLER) ≤ 10^{-2} (for a block size of 10^3 bits) and bit error ratio (BER) ≤ 10^{-5}. Whether BLER or BER is more meaningful depends upon the application. For transfers of large amounts of data with retransmittal protocols, BLER is appropriate. For facsimile applications and asynchronous character transfer, BER is more meaningful. The BER is usually the more stringent criterion. Note that typical switched network performance is an order of magnitude better than the above limiting criteria [11].

A.5 Performance criteria versus asynchronously connected G.726 32-kbit/s ADPCM links

A summary of test results that shows the maximum number of asynchronously connected G.726 32-kbit/s ADPCM links in tandem that enable the end-to-end performance criteria to be met is given in table A.1. Information regarding this data is provided in A.5.1 through A.5.4 and further details may be found in the documents cited in annex C, Bibliography.

A.5.1 Voice

For voice, four asynchronously connected G.726 32-kbit/s ADPCM links in tandem correspond to a value of Q = 20 dB. However, recognizing that there are other sources of distortion in a connection, such as additional PCM conversions, and to allow some margin for this eventuality, the standard provides a limit of at most three links for

voice applications. The limit of four G.726 32-kbit/s ADPCM links is reasonable if there are no other sources of significant distortion in the connection, and particularly if the connection contains no echo control devices.

A.5.2 Dual Tone Multi frequency (DTMF)

The range of encodings for DTMF is due to receiver sensitivity. In general, end-to-end network applications should employ receivers providing satisfactory performance through four asynchronously-connected G.726 32-kbit/s ADPCM links. Some older receivers embedded in the public network may only perform satisfactorily through two G.726 32-kbit/s ADPCM links. However, these receivers should tend to encounter fewer G.726 32-kbit/s ADPCM links in actual connections. The Category 1 signal allocation of one for the EC, IC and EC portions, respectively, combined with the typical uses of DTMF, will allow DTMF to be treated typically as a Category 1 signal.

A.5.3 Facsimile

Results for facsimile applications are not shown in table A.1. The criterion in this case is, of necessity, subjective. Results in Daumer and Sparrell [12] indicate that Groups I and II facsimile generally can be treated as Category 1 signals. Group III facsimile generally will be considered to be a Category 2 signal.

A.5.4 Voiceband data

The lower voiceband data encoding limits in table A.1 tend to correspond to the more stringent BER criteria. The results in table A.1 are based on the characteristics of specific modems as discussed by Daumer and Sparrell, and Kalb [12, 14]. More stringent requirements may apply to other types of modems, particularly if secondary or tertiary channels are used. The evaluation by Daumer and Sparrell [12] showed that one 4.8-kbit/s modem failed the criterion of BLER ≤ 10^{-2} through four G.726 32-kbit/s ADPCM links (the tested BLER was $1.03 \infty 10^{-2}$). The same modem was not evaluated by Kalb [14], but the results in [14] consistently showed that the BER criterion allowed one fewer encoding than the BLER criterion. Therefore, it was concluded that the same modem, at best, could meet only marginally the BER criterion through three asynchronously connected G.726 32-kbit/s ADPCM links. On this basis, some modems for specific applications may tolerate only two G.726 32-kbit/s ADPCM links at 4.8 kbit/s. The V.32 9.6 kbit/s modems can tolerate two G.726 32-kbit/s ADPCM links using the BLER criterion, but cannot tolerate any G.726 32-kbit/s ADPCM links with acceptable BER performance [8]. V.29 9.6 kbit/s modems, or similar modems, do not meet either the BER or BLER criterion at 9.6 kbit/s through any G.726 32-kbit/s ADPCM links, even with no analog impairments in the connection [7]. There is evidence that additional G.726 32-kbit/s ADPCM links in a connection increase the error ratio experienced [13]. The magnitude of the increase, which may be appreciable, depends upon the particular modem, the rate of transmission and the number of G.726 32-kbit/s ADPCM links already in the connection.

Voiceband data signals that function satisfactorily through three or more G.726 32-kbit/s ADPCM links should be treated as Category 1 signals. Data in table A.1 indicate that common modems such as those described above, operating at rates below 4.8 kbit/s, generally may be considered to be in Category 1; common modems operating at rates higher than 4.8 kbit/s generally may be considered to be in Category 2. Based on the test results and the BER and BLER criteria, common modems operating *at* 4.8 kbit/s generally may be considered to be in Category 1. In practice, there will be cases in which these signals will require consideration as being in Category 2, based on modem type, application and end-user expectations of satisfactory performance.

In general, the voiceband data results are based on an evaluation of several documents cited as normative references in the standard, and as informative references in clause A.2 and in annex C, Bibliography. All of this information leads to a consistent conclusion if both the BER and BLER criteria are used. The studies presented by Daumer and Sparrell, and Kalb [12,14] are of particular interest since they are based on a clearly defined reference connection derived from a characterization of a PSTN. The level of analog impairments in the total reference connection was derived from the 1982/83 End Office Connection Study (EOCS) [11]. An 85-percent impairment level was assumed for each analog parameter. This implies that some percentage of connections (less than 15 percent) would have poorer analog impairment levels and thus provide poorer end-to-end performance. This is consistent with predivestiture Bell System use of the BER and BLER criteria in that the error limits for 4.8 kbit/s maximum were supported on 85 percent of the completed connections with modems and circuit facilities provided by the Bell System.

Annex B

(informative)

Functional usage guidelines

B.1 General

This annex is provided as an aid in managing the use of G.726 32-kbit/s ADPCM technology to achieve the tandem encoding limits specified in the standard for either Category 1 or Category 2 signals.

Both PSTN and private-network applications are addressed. It is assumed that the only G.726 32-kbit/s ADPCM links in the connection occur in the EC and IC networks, and no allocation is made for the use of G.726 32-kbit/s ADPCM on the end-user's side of the network interface (NI). If G.726 32-kbit/s ADPCM is used on the end-user's side of the NI, the overall limits should be observed in order to achieve satisfactory performance.

B.2 Private-network connections

Figure B.1 illustrates the component portions of a typical private-network connection. For Category 1 signals, each of the three portions (the first EC portion, the IC portion and the second EC portion) should contain at most one G.726 32-kbit/s ADPCM link. Each EC portion extends from the NI to the IC's point of termination (POT) and, in general, the two EC portions may be provided by different ECs. In each EC portion, the EC responsible for providing service should limit the number of G.726 32-kbit/s ADPCM links to one. It is the IC's responsibility to ensure that the total IC portion contains no more than one G.726 32-kbit/s ADPCM link. When more than one IC is providing the IC portion, the overall IC service provider has the responsibility for meeting the overall IC portion G.726 32-kbit/s ADPCM tandem encoding limit. In summary:

1) For Category 1 signals, ECs should provide voice grade special access services that include no more than one 32-kbit/s ADPCM link;

2) For services supporting signals in Category 2, each of the three portions must have zero 32-kbit/s ADPCM links;

3) Special voice grade access services that could support signals in Category 2 should have no 32-kbit/s ADPCM links. ECs and ICs should cooperate in identifying service offerings that support signals in Category 2.

B.3 Public switched telecommunications network (PSTN) connections

The application of the G.726 32-kbit/s ADPCM tandem encoding limits to PSTN arrangements is similar to that for private networks. The guidelines given in B.2 for Category 1 and Category 2 signal cases are applicable to the IC's PSTN portion, but there are additional considerations for the EC portions.

Figure B.2 illustrates a typical PSTN connection. Each EC portion has two separate components:

1) The end-user's service connecting to the end office (EO). This frequently is basic residential or business exchange service, but could be any service providing PSTN connections at an EO;

2) The trunk-terminated switched-access arrangement from the EO to the POT.

The following guidelines apply to the EC portions of the connection:

1) For Category 1 signals, the EC portion, from NI to POT, should contain at most one G.726 32-kbit/s ADPCM link;

2) If services supporting signals in Category 2 are to contain zero G.726 32-kbit/s ADPCM links in the EC portion, both the end-user's service to the EO and the trunk-terminated switched-access arrangement must contain zero G.726 32-kbit/s ADPCM links. ECs should provide connections with zero G.726 32-kbit/s ADPCM links from the IC POT to each EO requested by the IC. In addition, ECs should cooperate with end-users and

their agents in identifying the service options available to obtain a path with zero G.726 32-kbit/s ADPCM links between the NI and the EO. Many end-user services may be accommodated by providing a type of data option on the end-user's service.

Annex C
(informative)

Bibliography

From Conference Record of the IEEE International Conference on Communications, Seattle, WA, June 1980[10)]

Cavanaugh, J. R., *A model for the subjective effects of quantizing distortion in digital transmission of speech,* pp. 20.5.1–20.5.5

From GLOBECOM 1985 Conference Record, IEEE, December 1985 [8)]

Ade, C. N.; Aprille, C. J.; Crandell, J. H.; Kramer, E. J.; Vera, A. J.; Warren, P. G., *32-KB/s ADPCM in the Western Electric D4 digital channel bank.* 37.3, pp. 1128–1132

Banerjee, S.; Feder, H.; Kilm, T.; Sparrell, D., *32-kbit/s ADPCM algorithm and line format standard for U. S. networks*. 37.6, pp. 1143–1147

Charbonnier, A.; Bonnet, J. M.; Picel, Z., *Performance evaluation of the CCITT 32 kbit/s ADPCM algorithm test sequences*. 37.2, pp. 1123–1127

Matsumara, T.; Gambe, H.; Murano, K.; Ohhata, M., *Implementation of CCITT standard 32 kbps ADPCM codec*. 37.1, pp. 1118–1122

Sakaguchi, H.; Morimura, Y.; Harakawa, K.; Maruta, R., *An Automated 32 kbps ADPCM Transcoder*. 37.5, pp. 1138–1142

From GLOBECOM 1984 Conference Record, IEEE, November 1984[11)]

Daumer, W. R.; Mermelstein, P.; Maitre, X.; Tokizawa, I., *Overview of the ADPCM coding algorithm*. 23.1, pp. 774–777

Kilm, T.; Sparrell, D., *Introduction of 32 Kb/s ADPCM equipment into the North American network*. 23.6, pp. 803–807

Psimenatos, N.; Gruber, J.; Goddard, G.; Mondor, D., *The application of 32 kb/s ADPCM systems in telecommunications networks*, pp. 796–802

Raulin, J. M.; Belfield, W. R.; Nishitani, T., *Objective test results for the 32 kb/s ADPCM coder*. 23.3, pp. 786–790

Taka, M.; Maruta, R; LeGuyader, A., *Synchronous tandem algorithm for 32 kbit/s ADPCM*. 23.4, pp. 791–795

Williams, G.; Suyderhoud, H*., Subjective performance evaluation of the 32-kbit/s ADPCM algorithm*. 23.2, pp. 778–785

ITU-T References[2)]

SG XVIII – Report R24, Source: ITU-T Secretariat, Report of the Meeting of Working Party XVIII/2 (Speech Processing), June 1983

SG XVIII – Report R28, Source: ITU-T Secretariat, Report of the Meeting of Working Party XVIII/2 (Speech Processing), December 1983

T1Y1 contributions, 1985–1986[6)]

Banerjee, S.; Schweizer, R., *ADPCM performance issues*. Document T1Y1.2/85-083, July 1985

Rondeau, D., *Comparison of 15 and 16 level ADPCM quantizing*. Document T1Y1.2/85-060, May 1985

Song, M.; Venkatataman, S.; Narasimha, M., *Performance evaluation of modified algorithms.* Document T1Y1.2/85-007, February 1986

1) In this standard, the abbreviation, G.726 32-kbit/s ADPCM, refers explicitly to the 32-kbit/s ADPCM algorithm as specified in ITU-T Recommendation G.726, *40, 32, 24, 16 kbit/s adaptive differential pulse code modulation (ADPCM)*, Geneva, 1990.

2) ITU-T is the current designation that replaced CCITT in February 1993. Available from American National Standards Institute, 11 West 42nd Street, New York, NY 10036.

3) IEEE P743 is considering the standardization of a method for detecting the presence of ADPCM on a connection.

4) It is likely that bit error ratios $\leq 1 \infty 10^{-8}$ will not degrade perceivably the synchronous tandeming property of ADPCM links. However, additional work is required to confirm this.

5) Available from University Microfilm International, 300 N. Zeeb Road, Ann Arbor, MI 48106.

6) Available from Alliance for Telecommunications Industry Solutions, 1200 G Street, NW, Suite 500, Washington DC 20005.

7) Available from COMSAT Corporation, Records Department, 22300 COMSAT Drive, Clarksburg, MD 20871.

10) Available from Ask IEEE, Attn: Manager Document Delivery, P.O. Box 4327, Burlingame, CA 94011-4327 (IEEE Catalog No. 87CH2424-0).

11) Available from Ask IEEE, Attn: Manager Document Delivery, P.O. Box 4327, Burlingame, CA 94011-4327 (IEEE Catalog No. 84CH2064-4).