American National Standard

for telecommunications –

network performance – tandem encoding limits for 32 kbit/s adaptive differential pulse-code modulation (ADPCM)

ANSI T1.501-1988

american national standards institute, inc.
1430 broadway, new york, new york 10018

American National Standard

Approval of an American National Standard requires verification 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 toward 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CAUTION NOTICE: This American National Standard may be revised or withdrawn at any time.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.

Published by

American National Standards Institute
1430 Broadway, New York, New York 10018

Printed in the United States of America

A1M1088/12

ANSI®
T1.501-1988

American National Standard
for Telecommunications –

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

Secretariat

Exchange Carriers Standards Association

Approved April 28, 1988

American National Standards Institute, Inc

Abstract

This standard specifies the limitations on the maximum number of 32 kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM) encodings, as described in ANSI T1.301-1987, 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.

Foreword

(This Foreword is not part of American National Standard T1.501-1988.)

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. ADPCM is one of a group of emerging technologies, each of which provides the capability for more efficient usage 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 ADPCM algorithm as specified in American National Standard for Telecommunications—Digital Processing of Voice-Band Signals—Algorithm and Line Format for 32 kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM), ANSI T1.301-1987. It is intended to provide for 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 adequately characterize 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.

Suggestions for improvement of this standard will be welcome. They should be sent to the Exchange Carriers Standards Association, Suite 200, 5430 Grosvenor Lane, Bethesda, MD 20814-2122.

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:

Ian M. Lifchus, Chair
Ivor N. Knight, Vice-Chair
O. J. Gusella, Jr, Secretary

Organization Represented	*Name of Representative*
EXCHANGE CARRIERS	
Alltel Service Corporation	John B. Kohut
Ameritech Services, Inc	Timothy D. Timmons Robert Koren (Alt)
Bell Atlantic	Salvatore J. Barbera Roger Nucho (Alt)
Bellcore	Ian M. Lifchus Martin T. Sullivan (Alt)
BellSouth Services	Leonard Strickland, Jr Hal B. Holton (Alt)
Centel Corporation	Joseph J. Olejar James W. Weith (Alt)
Contel Service Corporation	Russell G. DeWitt Stephen P. Welsh (Alt)
Exchange Carriers Standards Association	Joseph Mendoza James Eitel (Alt)
GTE SC/Telephone Operations	Greg L. Theus Richard L. Cochran (Alt)
National Telephone Cooperative Association	Joseph M. Flanigan
NYNEX	J. Fey Leo Katz (Alt)

Organization Represented	*Name of Representative*
Pacific Bell	Fred Doell
	Tom Garcia (Alt)
Puerto Rico Telephone Company	Segundo Ruiz
Southern New England Telephone	David M. Mangini
	David B. Montgomery (Alt)
Southwestern Bell Telephone Company	Joseph Mendoza
	Bob Caskey (Alt)
Telephone & Data Systems	Rudolph E. Hornacek
United States Telephone Association (USTA)	Paul K. Hart
	Thomas Gajeski (Alt)
United Telephone System, Inc	Harold Fuller
	David A. Eicher (Alt)
U S WEST	James Eitel
	Paul Hughes (Alt)
INTEREXCHANGE CARRIERS	
AT&T Communications	Henry L. Marchese
	Gerald H. Peterson (Alt)
CNCP Telecommunications	Roger M. Hay
	John D. Rogers (Alt)
Communications Transmission, Inc	Walter G. Lammert
	Stephen D. Klingler (Alt)
COMSAT	Ivor N. Knight
	Stephen J. Engelman (Alt)
Contel ASC	Michael K. Ward
	Raj Koneru (Alt)
MCI Telecommunications Corporation	Shekhar Tiwari
	Warren Liss (Alt)
Telecom Canada	B. Houle
	J. A. Zebarth (Alt)
Tymnet, Inc/McDonnell Douglas Corporation	Jeff Stern
	Rick Sprenkel (Alt)
US Sprint	Peter J. May
	Eric Scace (Alt)
Western Union Telegraph Company	Jacob Rabinowitz
MANUFACTURERS	
ADC Telecommunications, Inc	Patrick Cameron
	Nick Stanley (Alt)
Alcatel N. A.	Kenneth P. Ray
	Stig Persson (Alt)
AMP Inc	Edward Kelly
	George Lawrence (Alt)
AT&T Technologies	L. A. Hohmann
	J. R. Myrick (Alt)
Digital Equipment Corporation	Greg Borton
	Robert Di Meo (Alt)
DSC Communications Corporation	Allen Adams
	Kishan Shenoi (Alt)
ECI Telecom Inc	J. R. Kennedy
	Ron Murphy (Alt)
Ericsson, Inc	David Breeding
	Guy Campbell (Alt)
Exar Corporation	Mike Khan
	David Wong (Alt)
Fortel Corporation	Stephen V. Lyle
Fujitsu America, Inc	A. Bailey
	S. Minneman (Alt)
General DataComm Industries	Frederick Cronin
	Thomas Thompson (Alt)
GTE Communication Systems Corporation	Edward Glenner
	N. J. E. Reynolds (Alt)
Harris Corporation	Brian Jones
	David Cox (Alt)
Hekimian Laboratories, Inc	Larry E. Keith
	Mike F. Toohig (Alt)
Hewlett-Packard	Don C. Loughry
	Richard van Gelder (Alt)
Honeywell Bull	Robert F. Stover
IBM Corporation	Robert M. Amy
	Ronald C. Kunzelman (Alt)

Organization Represented	*Name of Representative*
Integrated Technology, Inc	Mark Lanning
	Dave Hensley (Alt)
LICOM, Inc	Lynn Chapman
	Maynard Graden (Alt)
M/A-COM Corporation	Michael Sun
Mitel Corporation	Keith Richardson
	Peter Perry (Alt)
Motorola, Inc	G. David Forney, Jr
	David Morgan (Alt)
Myriad Concepts, Inc	Michael Kelly
NCR Corporation	Thomas W. Kern
	Frank B. Andrews (Alt)
NEC America, Inc	Wayne Lohman
	Art Graham (Alt)
Northern Telecom Inc	David R. Cairns
Phoenix Microsystems, Inc	Robert E. Gewin
	Mark Hoffman (Alt)
Plantronics, Inc	M. Farrant
	Brian Cole (Alt)
Racal-Milgo, Inc	C. R. Baugh
	J. Jayapalan (Alt)
Rockwell International Corporation	James T. Carter, Jr
	Everett Turvey (Alt)
Siemens Communication Systems, Inc	Michael A. Pierce
	Dan R. Mondor (Alt)
Stromberg-Carlson Corporation	Roger Lanham
	Ron Kandell (Alt)
Telco Systems, Inc	Paul D. Lazay
	H. A. Carnes (Alt)
Telecommunications Techniques Corporation	Joseph A. Sciulli
	Bernard E. Worne (Alt)
Telex Computer Products, Inc	Frederick Skoog
	Anne Davis (Alt)
TELINQ Systems, Inc	Jeff Foust
	Robert Felice (Alt)
TIMEPLEX, Inc	Shiv Verma
	Paul Lue (Alt)
Unisys	Stanley D. Fenner
	Marvin Bass (Alt)
U.S. Telecommunications Suppliers Association	Michael J. Birck
	Paul DeLuca (Alt)
Verilink Corporation	William J. Buckley
Videoconferencing Systems, Inc	John Nuwer
	Ken Hutchison (Alt)
GENERAL INTEREST	
American Broadcasting Companies, Inc	Howard Meiseles
	Ken Michel (Alt)
Ashford Associates	Donald A. Ashford
Association of Telemessaging Services International, Inc (ATSI)	Joseph N. Laseau
	Mark J. Golden (Alt)
BDM Corporation	Roy Bernd
	Douglas Zolnick (Alt)
Carter Hawley Hale-Information Services	James A. Rothenberger
	Vic Accettura (Alt)
CBS Broadcasting Group, CBS, Inc	Stavros Hilaris
	Joseph McNulty (Alt)
Creative Communications Consulting	Richard Bobilin
	James Boe (Alt)
Defense Communications Agency	Martin A. Thompson
	Benjamin E. Bakke (Alt)
Federal Express Corporation	Michael E. Varrassi
	Paul L. Davis (Alt)
Martin-Marietta Information and Communications Systems	J. D. McCreary
	Mike Maher (Alt)
National Broadcasting Company	Ronald J. Gnidziejko
National Bureau of Standards	Robert Rountree, Jr
National Communications System	Charles D. Bodson
	George W. White (Alt)

Organization Represented	*Name of Representative*
National Telecommunications and Information Administration/ Institute for Telecommunication Sciences (NTIA/ITS)	William F. Utlaut
	Neal B. Seitz (Alt)
OMNICOM, Inc	Harold C. Folts
Rural Electrification Administration	M. Wilson Magruder
	Gerald S. Schrage (Alt)
US General Services Administration	Thomas J. Drury
	William C. Rinehuls (Alt)
Utilities Telecommunications Council	Frank S. Law
	Shirley S. Fujimoto (Alt)

At the time it approved this standard, Technical Subcommittee for Performance, T1Q1, had the following members:

Arthur K. Reilly, Chair
Melvin N. Woinsky, Vice-Chair
Richard J. Schweizer, Secretary

Organization Represented	*Name of Representative*
ADC Telecommunications, Inc	Uday Kabe
	John Frederick (Alt)
Amdahl Communications, Inc	Farooq Raza
American Broadcasting Companies, Inc	Howard Meiseles
	Ken Michel (Alt)
Ameritech Services	Peter K. Cencer
	Raymond W. Norton (Alt)
AT&T Communications	Nerses Kazanjian
	Melvin N. Woinsky (Alt)
AT&T Technologies	Richard Murphy
	R. E. Langseth (Alt)
Bell Atlantic	Duey Patrick
	Harry Hetz (Alt)
Bellcore	Ralph E. Jensen
	Thomas C. Spang (Alt)
BellSouth Services	H. B. Holton
	H. C. Irby (Alt)
Canadian Broadcasting Corporation	Brian Sawyer
CBS, Inc	Stavros Hilaris
	Joseph M. McNulty (Alt)
COMSAT	Mark T. Neibert
	Ivor N. Knight (Alt)
Creative Communications Consulting	Richard T. Bobilin
	James Boe (Alt)
DB TELCO	Larry S. DiBiaso
	C. R. Crane (Alt)
Defense Communications Agency	Granger Kelley
	Walter Cybrowski (Alt)
DSC Communications Corporation	Paul Yang
	Phil Snow (Alt)
Ericsson, Inc	Lori A. Kipp
	Bengt Lagerstedt (Alt)
Fortel Corporation	Stephen V. Lyle
General DataComm Industries, Inc	William Potter
	David Moon (Alt)
GTE SC/Telephone Operations	Michael L. Kanotz
	B. J. Harris (Alt)
Lynch Communication Systems, Inc	Benjamin M. Rice
	Bernard Kwok (Alt)
M/A-COM Corporation	Hossein Besharatian
	David Castiel (Alt)
MCI Telecommunications Corporation	M. Whiting Thayer
	Peter Wallingford (Alt)
Motorola, Inc	Curtis W. Dodd
	Paul Nelson (Alt)
National Broadcasting Company, Inc	Ronald J. Gnidziejko
National Communications System	Neal B. Seitz

Organization Represented	*Name of Representative*
National Telecommunications and Information Administration/ Institute for Telecommunication Sciences (NTIA/ITS)	Neal B. Seitz
	Edmund A. Quincy (Alt)
NEC America, Inc	Takunori Ito
	T. Takabatake (Alt)
Northern Telecom, Inc	David R. Cairns
	Godfrey Williams (Alt)
NYNEX	J. J. McDonough
	P. A. Campanella (Alt)
Pacific Bell	H. M. Hart
	Jim Weibert (Alt)
Phoenix Microsystems, Inc	Joseph Sturbois
	Rae Brockman (Alt)
Rockwell International	J. A. Crossett
	Joe Russo (Alt)
Siemens Communication Systems, Inc	John E. Whitecar
	Pierre Sibille (Alt)
Southern New England Telephone	A. Ralph Day
Southwestern Bell Telephone Company	Bill Litzinger
	Bob Hall (Alt)
STARNET International, Inc	Richard H. Fish
Telecom Canada	B. Sambasivan
	W. C. Parker (Alt)
TIMEPLEX, Inc	Paul Lue
	Shue-Yu Kwan (Alt)
United Telephone System, Inc	David A. Eicher
	David M. Emerick (Alt)
US Sprint	Cannon Hwu
	Peter J. May (Alt)
U S WEST Bell Tri-Co Services	James L. Eitel
	L. D. Hofer (Alt)
Utilities Telecommunications Council	Steve F. Lux
Western Union Telegraph Company	Jacob Rabinowitz

Working Group T1Q1.1, 4-kHz Voice and Voiceband Data, which developed this standard, had the following officers and participants:

Godfrey Williams, Chair
Peter K. Cencer, Vice-Chair
Nerses Kazanjian, Secretary

Dick Bobilin
Ernest L. Carter
Ted L. Church
Robert Creighton
Ralph Day
Larry DiBiaso
Curtis W. Dodd
Jack Douglass
Sally Dunne
Chuck Dvorak
Marty Egan
Ferial El-Mokadem
Edmond Elowe
Ken Evans
K. E. Ferguson
Harold Ferrell
Todd Halvorson
Susan Hanson
Howard Hart
L. D. Hofer
Hal B. Holton
Allan Hoosain
Cannon Tsong-Min Hwu
James Ingle
Al Jaakkola
Ralph E. Jensen
Michael Kalb
Michael L. Kanotz
Ross Keating
Bob Koester
Bengt Lagerstedt
Alvin Lai
Steven Lux
Esmat Mahmoud
B. Manseur
Peter J. May
John J. McDonough
Phil Meanor
H. S. Merrill, Jr
Steve Minneman
David Moon
Jack Moran
Richard Murphy
Farzad Nabavi
Andrew Niedzwieki
Donald C. O'Connor
Bill Potter
Arthur K. Reilly
Steve Sackel
Bob Schiff
Richard J. Schweizer
Neal B. Seitz
Steven Sheldon
Thomas C. Spang
Robert Stebbins
Richard Taffet
Whitey Thayer
Michael E. Vitella
Robert Waller
James M. Weibert
Michael Weston
Melvin N. Woinsky
Paul Yang
Yoshihiko Yokoyama
Larry Young
J. A. Zebarth

Contents

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. The standard covers the use of 32-kbit/s Adaptive Differential Pulse-Code Modulation (T1.301 ADPCM)[1] in 4-kHz voicegrade network connections and limits the number of T1.301 ADPCM links allowed. It relates to the transport of signals processed by the 32-kbit/s T1.301 ADPCM algorithm as specified in American National Standard for Telecommunications—Digital Processing of Voice-Band Signals—Algorithm and Line Format for 32 kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM), ANSI T1.301–1987. 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 T1.301 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 voicegrade connections established on either public-switched telephone networks (PSTNs) or private-networks. The tandem encoding limits of this standard apply to exchange carriers (ECs), interexchange carriers (ICs), and private-network providers.

1. In this standard, the abbreviation, T1.301 ADPCM, will refer explicitly to the 32-kbit/s ADPCM algorithm as specified in American National Standard for Telecommunications—Digital Processing of Voice-Band Signals—Algorithm and Line Format for 32 kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM), ANSI T1.301-1987.

2. Referenced Standards and Publications

2.1 Referenced American National Standard. This standard is intended to be used with American National Standard for Telecommunications—Digital Processing of Voice-Band Signals—Algorithm and Line Format for 32 kbit/s Adaptive Differential Pulse-Code Modulation (ADPCM), ANSI T1.301–1987.

2.2 Other Referenced Standard. This standard is also intended to be used with CCITT Recommendation G.113, Transmission Impairments, Red Book, Vol. III, Fascicle III.1.[2]

2.3 Referenced Publications

[1] Cavanaugh, J. R. A model for the subjective effects of quantizing distortion in digital transmission of speech. In: International Conference on Communications. Seattle, Washington. June 8–12, 1980. pp. 20.5.1–20.5.5.

[2] Psimenatos, N.; Gruber, J.; Goddard, G.; Mondor, D. The application of 32 kb/s ADPCM systems in telecommunications networks. In: IEEE Globecom 1984 Conference Record. New York: IEEE. 1984 November. pp. 796–802.

3. Definitions

adaptive differential pulse-code modulation (ADPCM). A digital transcoding technique in

2. Available from the American National Standards Institute, 1430 Broadway, New York, NY 10018.

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.

ADPCM link. Any transmission system, or part of a transmission system, which performs one unique PCM–ADPCM transcoding and a corresponding ADPCM–PCM transcoding.

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

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.

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.

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

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.

4. Specification of Limits

4.1 General. Traditional analog transmission parameters used in characterizing the performance of the various segments of end-to-end connections are not adequate to retain quality transmission performance if T1.301 ADPCM equipment is deployed freely in the various segments. Currently no standard test procedure is available to quantify the effect of T1.301 ADPCM links in a connection or part thereof. Therefore, the only method available to control the transmission performance impact of T1.301 ADPCM is to limit the number of T1.301 ADPCM links in connections. Information regarding the technical basis for the limits specified in this standard is included in Appendix A, ADPCM Performance Effects.

4.2 Categories of Signals. The limitation on T1.301 ADPCM links in public or private networks depends upon the signal type being transmitted over network connections. Appendix A provides information describing the effects of T1.301 ADPCM transcoding on several common signal types. The various signal types conveyed by voicegrade 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 T1.301 ADPCM links.

Category 2. Voiceband (4-kHz) signals that cannot be transmitted with satisfactory performance over connections containing T1.301 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 T1.301 ADPCM tandem encoding limits set forth in this standard, the following criteria were used:

(1) *Voice.* The criterion is based on a measure called Q that captures the effect of quantizing noise on quality[1].[3] The criterion of $Q \geq 20$ dB was used.

(2) *Voiceband Data.* Two criteria were used: Block Error Ratio (BLER) $\leq 10^{-2}$ (with a block size of 10^{-3} bits) and Bit Error Ratio (BER) $\leq 10^{-5}$. 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.

(3) *Dual-Tone Multifrequency (DTMF) signals.* The criterion of digit error ratio $< 10^{-3}$ was used.[2]

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

3. Numbers in brackets correspond to numbers in 2.3, Referenced Publications.

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 synchronously connected T1.301 ADPCM links is equivalent (except for delay) to one link if (1) the transmission of the 32-kbit/s T1.301 ADPCM links and the intermediate 64-kbit/s PCM links is error free[4] and (2) the 32-kbit/s T1.301 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 cancellers or digital loss pads, will not satisfy the above constraints. If two T1.301 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 T1.301 ADPCM link and the performance of two asynchronously connected T1.301 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 T1.301 ADPCM links.

4.4 Tandem Encoding Limits. The total number of T1.301 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 effective number of asynchronously connected T1.301 ADPCM links in these network connections shall be limited as follows:

(1) *Category 1 Signals.* The limit is three asynchronously connected T1.301 ADPCM links.

(2) *Category 2 Signals.* The limit is zero T1.301 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 this case, greater flexibility is possible, in terms of the number of T1.301 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 Appendix 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 T1.301 ADPCM is not deployed on the end-user's side of the network interface (NI). If T1.301 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 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 T1.301 ADPCM links. See Appendix B, Functional Usage Guidelines for implementation information.

For both private and public switched connections, if an EC provides a T1.301 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 T1.301 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 T1.301 ADPCM links in the national extension. The allocation specified in the provisional planning rule, as described in CCITT Recommendation G.113, allows at most one T1.301 ADPCM link in the national extension. However, an exception is noted that would allow at most two T1.301 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.

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

Appendixes

(These Appendixes are not part of American National Standard T1.501–1988, but are included for information only.)

Appendix A
ADPCM Performance Effects

A1. General

The impact of T1.301 ADPCM on the performance of a voicegrade 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 T1.301 ADPCM links in the connection. This Appendix is provided as an aid to the reader in understanding the limits in the standard. The information in this Appendix was derived from evaluations of specific modems and DTMF receivers of several manufacturers and may not describe performance of all available similar hardware.

A2. References

The following documents are referenced in this Appendix:

[A1] CCITT Recommendation G.711, Pulse Code Modulation (PCM) of Voice Frequencies. Red Book, Volume III - Fascicle III.3. Geneva, 1972.[2]

[A2] CCITT Recommendation G.721, 32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM). Red Book, Volume III - Fascicle III.3. Malaga-Torremolinos, 1984.[2]

[A3] CCITT Recommendation G.721, 32 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM) (revised) Geneva, 1986.[2]

[A4] Daumer, W. R.; Sparrell, D. "FSK Enhancements for the Proposed T1Y1 ADPCM Algorithm," Document T1Y1.2/86-006, January 1986.[5]

[A5] SG XVIII - Report R26, Source: CCITT Secretariat, "Report of the Meeting of Working Party XVIII/8 (Speech Processing)," July 1986.

[A6] CCITT Recommendation, G.113, Transmission Impairments. Red Book, Vol. III - Fascicle III.1. Malaga-Torremolinos, 1984.[2]

[A7] T1Q1/85-025R1, "Response to T1X1.4 on 64 kb/s Clear Channel Capability," November 1985.

[A8] 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: 1984 November.

[A9] Kalb, M. ADPCM performance impact on voiceband data. In: IEEE Globecom 1985 Conference Record. New York: IEEE: December 1985. pp. 1133–1137.

[A10] CCITT 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. Red Book, Volume VIII - Fascicle VIII.1. Malaga-Torremolinos, 1984.[2]

[A11] CCITT Recommendation V.29, 9600 Bits Per Second Modem Standardized for Use on Point-to-Point 4-Wire Leased Telephone-Type Circuits. Red Book, Volume VIII - Fascicle VIII.1. Malaga-Torremolinos, 1984.[2]

A3. Definitions

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.

quantizing distortion unit (qdu). The distortion that is subjectively equivalent to the quantizing distortion introduced by an average

5. Available from Exchange Carriers Standards Association, 5430 Grovesnor Lane, Bethesda, MD 20814-2122.

8-bit codec pair (A/D and D/A conversion, A-law or μ-law) that complies with CCITT Recommendation G.711 [A1].[6]

A4. 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 CCITT 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 CCITT studies are based on the earlier CCITT algorithm [A2] rather than the current CCITT algorithm [A3] (which is identical to ANSI T1.301–1987). However, it has been shown that, except for the improved performance of the current algorithm with respect to frequency-shift keying modems, the performance of the current algorithm is essentially identical to that of the old algorithm [A4,A5]. The CCITT in Recommendation G.113 [A6] 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 tandemed 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 T1.301 ADPCM links in tandem. The additional round-trip delay consists of up to six T1.301 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 [A7].

For voiceband data, two criteria are used: Block Error Ratio (BLER) equal to or less than 10^{-2} (for a block size of 10^{3} bits) and Bit Error Ratio (BER) equal to or less than 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 [A8].

A5. Performance Criteria versus Asynchronously Connected T1.301 ADPCM Links

A summary of test results that shows the maximum number of asynchronously connected T1.301 ADPCM links in tandem that enable the end-to-end performance criteria to be met is given in Table A1. Information regarding this data is provided in A5.1 through A5.3 and further details may be found in the references given in Appendix C.

A5.1 Voice. For voice, four asynchronously connected T1.301 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

Table A1
Summary of Test Results
Number of T1.301 ADPCM Links in Tandem

	Number of T1.301 ADPCM Links (Application Dependent)	Reference/ Discussion Section
Voice	3-4*	A5.1
DTMF	2-4	A5.2
Voiceband Data		
$\leq$ 2.4 kbit/s	4	A5.3
4.8 kbit/s	2-4	A5.3
9.6 kbit/s	0	A5.3
V.32 9.6 kbit/s	0-2	A5.3

* Total additional round-trip delay, if there are no echo control devices in the connection, should be below 5 ms.

6. Numbers in brackets correspond to numbers in Section A2, References.

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 T1.301 ADPCM links is reasonable if there are no other sources of significant distortion in the connection, and particularly if the connection contains echo control devices.

A5.2 DTMF and Graphics. The range of encodings specified for DTMF is due to receiver sensitivity. In general, end-to-end network applications should employ receivers providing satisfactory performance through four asynchronously connected T1.301 ADPCM links. Some older receivers embedded in the public network may only perform satisfactorily through two T1.301 ADPCM links. However, these receivers should tend to encounter fewer T1.301 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 typically be treated as a Category 1 signal. Results for graphic applications are not shown in Table A1. The criteria in this case is of necessity subjective. Results in reference [A4] indicate that Groups I and II facsimile can generally be treated as Category 1 signals. Group III facsimile will generally be considered as a Category 2 signal.

A5.3 Voiceband Data. The lower voiceband data encoding limits in Table A1 tend to correspond to the more stringent BER criteria. The results in Table A1 are based on the characteristics of specific modems as referenced in the cited documentation, in particular references [A4] and [A9]. More stringent requirements may apply to other types of modems, particularly if secondary or tertiary channels are used. The evaluation in reference [A4] showed that one 4.8-kbit/s modem failed the criterion of BLER $\leq 10^{-2}$ through four T1.301 ADPCM links (the tested BLER was 1.03×10^{-2}). The same modem was not evaluated in [A9], but the results in [A9] 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 only marginally meet the BER criterion through three asynchronously connected T1.301 ADPCM links. On this basis some modems for specific applications may only tolerate two T1.301 ADPCM links at 4.8 kbit/s. The V.32-type 9.6-kbit/s modems can tolerate two T1.301 ADPCM links using the BLER criteria but cannot tolerate any T1.301 ADPCM links with acceptable BER performance [A10]. V.29-type 9.6-kbit/s modems, or similar modems, do not meet either of the BER or BLER criteria at 9.6 kbit/s through any T1.301 ADPCM links, even with no analog impairments in the connection [A11]. There is evidence that additional T1.301 ADPCM links in a connection may increase the error ratio experienced. The magnitude of the increase, which may be appreciable, depends on the particular modem, the rate of transmission, and the number of T1.301 ADPCM links already in the connection.

Voiceband data signals that satisfactorily function through three or more T1.301 ADPCM links should be treated as Category 1 signals. Data in Table A1 indicate that common modems such as those described above, operating at rates below 4.8 kbit/s, may generally be considered to be in Category 1; common modems operating at rates higher than 4.8 kbit/s may generally 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 may generally 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 contained in the bibliography (Appendix C). All of this information leads to a consistent conclusion if both the BER and BLER criteria are used. The studies contained in references [A4] and [A9] 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) [A8]. An 85% impairment level was assumed for each analog parameter. This implies that some percentage of connections (less that 15%) 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% of the completed connections with modems and circuit facilities provided by the Bell System.

Appendix B
Functional Usage Guidelines

B1. General

This appendix is provided as an aid in managing the use of T1.301 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 T1.301 ADPCM links in the connection occur in the EC and IC networks and no allocation is made for the use of T1.301 ADPCM on the end-user's side of the network interface (NI). If T1.301 ADPCM is used on the end-user's side of the NI, the overall limits should be observed in order to achieve satisfactory performance.

B2. Private-Network Connections

Figure B1 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 T1.301 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 T1.301 ADPCM links to one. It is the IC's responsibility to ensure that the total IC portion contains no more than one T1.301 ADPCM link. When more than one IC is providing the IC portion, the IC overall service provider has the responsibility for meeting the overall IC portion T1.301 ADPCM tandem encoding limit.

For services supporting signals in Category 2, each of the three portions must have zero T1.301 ADPCM links.

For Category 1 signals, ECs should provide voicegrade special access services that include no more than one T1.301 ADPCM link.

Special voicegrade access services that would support signals in Category 2 should have no T1.301 ADPCM links. ECs and ICs should cooperate in identifying service offerings that support signals in Category 2.

B3. Public Switched Telephone Network Connections

The application of the T1.301 ADPCM tandem encoding limits to PSTN arrangements is similar to that for private networks but with additional considerations for the EC portions. The guidelines given in B2 for both Category 1 and Category 2 signal cases are applicable to the IC's PSTN portion.

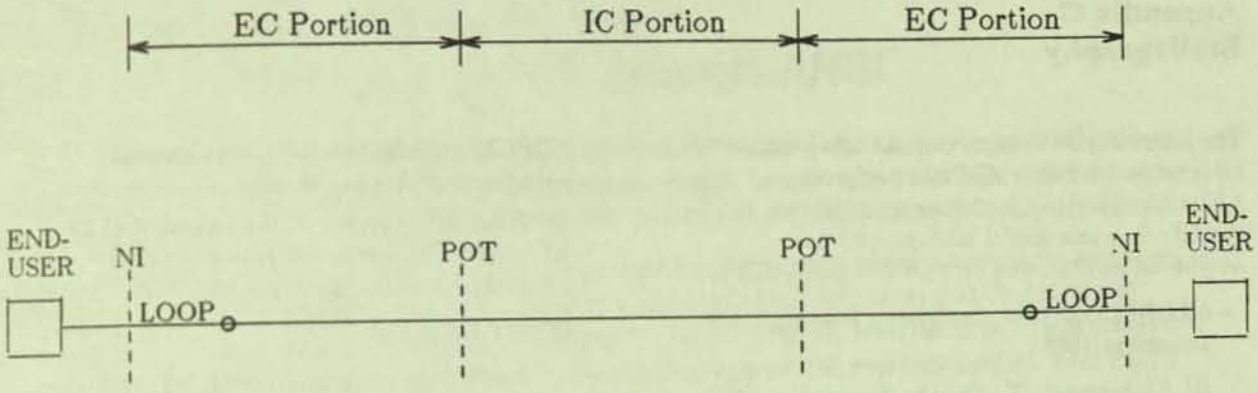
Figure B2 illustrates a typical PSTN connection. Each EC portion has two separate components:

(1) The end-user's service connecting to the end office (EO), which is frequently 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 POT to the EO.

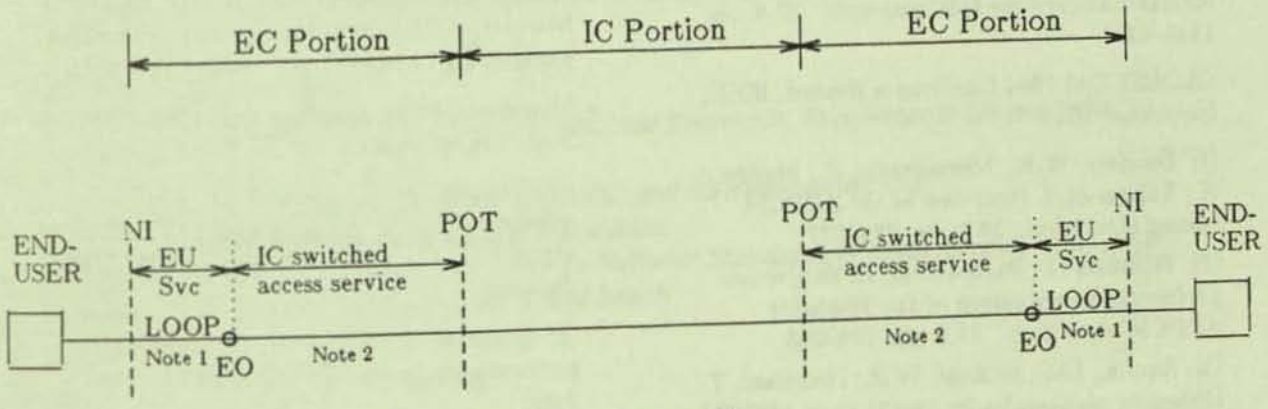
For Category 1 signals, the EC portion, from NI to POT, should contain at most one T1.301 ADPCM link.

For services supporting signals in Category 2 to have zero T1.301 ADPCM links in the EC portion, both the end-user's service to the EO and the switched access service must have zero T1.301 ADPCM links. ECs should provide connections with zero T1.301 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 T1.301 ADPCM links between the NI and the EO. Many end-user services may be accommodated by the EC's providing a type of data option on the end-user's service.



NOTES: IC = Interexchange carrier; EC = Exchange carrier; POT = Point of termination; NI = Network interface.

Figure B1
Private Network Connection



NOTES: (1) Any service providing PSTN connections to end-users at an EO.

(2) Trunk-terminated switched-access arrangement.

(3) EU = End-User; IC = Interexchange carrier; EC = Exchange carrier; POT = Point of termination; PSTN = Public switched telephone network; NI = Network interface.

Figure B2
Typical Public Switched Telephone Network Connection

Appendix C
Bibliography

The following documents contain information relevant to 32-kbit/s ADPCM performance. Although this list is by no means complete, it should serve as a useful bibliography for additional information on ADPCM performance.

- GLOBECOM 1985 Conference Record, IEEE, December 1985.

  [1] Matsumara, T.; Gambe, H.; Murano, K.; Ohhata, M. Implementation of CCITT standard 32 kbps ADPCM codec. 37.1. p. 1118–1122.

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

  [3] Ade, C.N.; Aprille, T.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.

  [4] Kalb, M. ADPCM performance impact on voiceband data. 37.4. pp. 1133–1137.

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

  [6] 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.

- GLOBECOM 1984 Conference Record, IEEE, November 1984.

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

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

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

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

  [5] Psimenatos, N.; Gruber, J.; Goddard, G.; Mondor, D. The application of 32 kb/s ADPCM systems in telecommunications networks. 23.5. pp. 796–802.

  [6] Kilm T.; Sparrell, D. Introduction of 32 Kb/s ADPCM equipment into the north american network. 23.6. pp. 803–807.

- CCITT References.

  [1] CCITT Recommendation G.721, 32 kbit/s Adaptive differential pulse code modulation (ADPCM). Red Book, Volume III - Fascicle III.3, Malaga-Torremolinos, 1984.

  [2] CCITT Recommendation G.721, 32 kbit/s Adaptive differential pulse code modulation (ADPCM) revised Geneva, 1986.

  [3] CCITT Recommendation G.113, Transmission Impairments, Connections and Circuits, Red Book, Volume III - Fascicle III.1, Malaga-Torremolinos, 1984.

  [4] SG XVIII - Report R28. Source: CCITT Secretariat, Report of the meeting of working party XVIII/2 (speech processing). 1983 December.

  [5] SG XVIII - Report R24. Source: CCITT Secretariat, Report of the meeting of working party XVIII/2 (speech processing). 1983 June.

  [6] SG XVIII - Report R26. Source: CCITT Secretariat, Report of the meeting of working party XVIII/8 (speech processing). 1986 July.

- Numerous T1Y1 contributions, 1985-1986. Some examples are:

  [1] Daumer, W. R.; Sparrell, D.; FSK enhancements for the proposed T1Y1 ADPCM algorithm. Document T1Y1.2/86-006. 1986 January.[5]

  [2] Banerjee, S.; Schweizer, R. ADPCM performance issues. T1Y1.2/85-083. 1985 July.[5]

  [3] Song, M.; Venkatataman, S.; Narasimha, M. Performance evaluation of modified algorithms. T1Y1.2/86-007. 1986 February.[5]

  [4] Rondeau, D. Comparison of 15 and 16 level ADPCM quantizing. T1Y1.2/85-060. 1985 May.[5]

Ready Access to the World's Standards through ANSI

Are you seeking a standard vital to your interests? Turn to the American National Standards Institute, which performs a unique function as America's clearinghouse and information center for national, international, regional, and foreign standards.

ANSI is the sole source of *all* approved American National Standards. In addition, the Institute is the U.S. source for all international standards and drafts of the International Organization for Standardization (ISO), International Electrotechnical Commission (IEC), and of the publications of ISO member bodies. And that's not all . . .

Here's a capsule summary of the wealth of materials available from ANSI —

STANDARDS

- *All approved American National Standards in print. A standing order service is available to standards users who wish to receive new and revised standards automatically.*
- *ISO and IEC International Standards*
- *ISO Draft International Standards*
- *IEC Six Months' Rule Drafts*
- *CEN Proposals — Draft European Standards developed and issued for preliminary vote by the European Committee for Standardization (CEN).*
- *CENELEC/CECC Proposals — Draft specifications issued by CECC, the Electronic Components Committee of the European Committee for Electrotechnical Standardization (CENELEC)*
- *Standards of 75 national standards organizations that belong to ISO*
- *English translations of thousands of foreign standards*

CATALOGS

- *Annual Catalog of American National Standards. Supplements are published several times a year.*
- *Latest catalogs issued by ISO, IEC, and 65 members of ISO*
- *Foreign catalogs in English translation*
- *Specialized listings of American National Standards in fields ranging from information processing to safety and health*

Available in Microform

American National Standards are available in microform from Information Handling Services (IHS), Inverness Business Park, 15 Inverness Way East, P.O. Box 1154, Englewood, Colorado 80150; tel (303) 790-0600. All ISO and IEC standards may also be obtained in microform from IHS.

American National Standards

The standard in this booklet is one of 8000 standards approved to date by the American National Standards Institute.

The Standards Institute provides the machinery for creating voluntary standards. It serves to eliminate duplication of standards activities and to weld conflicting standards into single, nationally accepted standards under the designation "American National Standards."

Each standard represents general agreement among maker, seller, and user groups as to the best current practice with regard to some specific problem. Thus the completed standards cut across the whole fabric of production, distribution, and consumption of goods and services. American National Standards, by reason of Institute procedures, reflect a national consensus of manufacturers, consumers, and scientific, technical, and professional organizations, and governmental agencies. The completed standards are used widely by industry and commerce and often by municipal, state, and federal governments.

The Standards Institute, under whose auspices this work is being done, is the United States clearinghouse and coordinating body for voluntary standards activity on the national level. It is a federation of trade associations, technical societies, professional groups, and consumer organizations. Some 1000 companies are affiliated with the Institute as company members.

The American National Standards Institute is the United States member of the International Organization for Standardization (ISO) and the International Electrotechnical Commission (IEC). Through these channels U.S. standards interests make their positions felt on the international level. American National Standards are on file in the libraries of the national standards bodies of more than 60 countries.

American National Standards Institute, Inc
1430 Broadway
New York, N.Y. 10018