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TRANSMISSION SYSTEMS AND MEDIA

GENERAL RECOMMENDATIONS ON THE TRANSMISSION QUALITY FOR AN ENTIRE INTERNATIONAL TELEPHONE CONNECTION

TRANSMISSION IMPAIRMENTS

ITU-T Recommendation G.113

(Previously "CCITT Recommendation")

FOREWORD

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

The World Telecommunication Standardization Conference (WTSC), which meets every four years, established the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

ITU-T Recommendation G.113 was revised by the ITU-T Study Group XII (1988-1993) and was approved by the WTSC (Helsinki, March 1-12, 1993).

NOTES

1 As a consequence of a reform process within the International Telecommunication Union (ITU), the CCITT ceased to exist as of 28 February 1993. In its place, the ITU Telecommunication Standardization Sector (ITU-T) was created as of 1 March 1993. Similarly, in this reform process, the CCIR and the IFRB have been replaced by the Radiocommunication Sector.

In order not to delay publication of this Recommendation, no change has been made in the text to references containing the acronyms "CCITT, CCIR or IFRB" or their associated entities such as Plenary Assembly, Secretariat, etc. Future editions of this Recommendation will contain the proper terminology related to the new ITU structure.

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

CONTENTS

TRANSMISSION IMPAIRMENTS

(Geneva, 1980; amended at Malaga-Torremolinos, 1984;
Melbourne, 1988 and Helsinki, 1993)

1 Transmission impairment

1.1 The objectives for the attenuation distortion of a maximum-length 4-wire chain are given in Recommendation G.132 and those of the signal-independent noise performance of such maximum-length connections are given in clause 2. Bearing in mind that less complicated connections (which are more numerous) will have less attenuation distortion and less noise, then the maximum, average and minimum values of loudness rating recommended in Recommendation G.121 will ensure an adequate transmission performance on international connections.

1.2 Should values of attenuation distortion or noise greatly differe from those recommended by the CCITT for systems and equipment be contemplated, then guidance concerning possible changes in transmission performance can be found in Recommendation P.11 and Annexes, with some indication of possible trade-offs between them.

2 Network performance objective for circuit noise on complete telephone connections

The CCITT recommends that the network performance objective for the mean value, expressed in decibels and taken over a large number of worldwide connections (each including four international circuits), of the distribution of one-minute mean values of signal-independent noise power of the connections, should not exceed –43 dBm0p referred to the input of the first circuit in the chain of international circuits.

3 Transmission impairments due to digital processes

The incorporation of unintegrated digital processes in international telephone connections, particularly during the mixed analogue/digital period, can result in an appreciable accumulation of transmission impairments. It is, therefore, necessary to ensure that this accumulation does not reach a point where it can seriously degrade overall transmission quality.

3.1 Quantizing distortion

From the point of view of quantizing distortion, it is recommended that no more than 14 units of quantizing distortion (qdu) or equivalent total code distorsion should be introduced in an international telephone connection.

For telephone connections which incorporate unintegrated digital processes, it is permissible to simply add the units of quantizing distortion that have been assigned to the individual digital processes to determine the total or overall quantizing distortion. Some sources of quantizing distortion and the units tentatively assigned to them are given in 3.2.

By definition, an average 8-bit codec pair (A/D and D/A conversions, A-law or μ-law) which complies with Recommendation G.711 introduces 1 quantizing distortion units (1 qdu). An average codec pair produces about 2 dB less quantizing distortion than the limits indicated in Recommendation G.712. This would correspond to a single-to-distortion ratio of 35 dB for the sine-wave test method and approximately 36 dB for the noise test method. (A total of fourteen 8-bit PCM processes each of which just comply with the limits for signal-to-distortion ratio in Recommendation G.712 would be unacceptable). The same principle should be applied when proposing planning values of quantizing distortion units for other digital processes.

In principle, the number of units for other digital processes are determined by comparison with an 8-bit PCM codec pair such that the distortion of the digital process being evaluated is assigned *n* quantizing distortion units if it is equivalent to *n* unintegrated 8-bit PCM process in tandem. Several methods of comparison are possible; these include objective measurements (or equivalent analysis), subjective tests, and data tests in which the effect on the bit error ratio at the output of a voice-band data modem receiver is used as a criterion.

At the present time no objective measurement capability exists which can produce results (e.g. SNR) that correlate closely with results obtained from subjective measurement of the effect of many of the digital processes now being studied on speech performance. Therefore, the number of units of quantization distortion for digital processes should, in general, be determined by subjective measurement methods, such as those found in Recommendation P.83. In some instances the number of units of quantization distortion for a digital process can be determined without subjective measurement by decomposing a digital process into two or more parts and allocating to the parts suitable fractions of the total number of units assigned to the digital process. However, while this method may be considered an objective method for determining the qdu assignments for the parts, it uses as a starting point a subjectively determined value. Furthermore, except for relatively simple digital processes where the decomposition is uncomplicated, this method may not be reliable and should be used with care.

Planning rules should be aplicable to all signals transmitted in the voice-frequency band. Therefore, in general, both speech quality and data performance must be considered. Speech quality should be evaluated by subjective tests and data performance should be evaluated by objective measurements which provide estimates of the expected bit error ratio and signalling performance. At present, however, because of the lack of an objective method for evaluating the effect of digital processes on voice-band data performance, the planning rule in this Recommendation is limited to voice connection planning purposes only. Clause 4 discusses some of the problems associated with developing a planning rule for connections carrying voice-band data and other non-speech signals. Such a rule would be based on a unit reflecting the contribution digital processes make to the impairment or impairments that affect voice-band data modems and/or signalling systems. Such a unit does not exist yet.

NOTE – qdu is defined in terms of quantizing distortion as present in PCM and other waveform coders and assumes that the quantizing distortion adds on a 15 $\log_{10}(n)$ law for *n* codec pairs in tandem. There is some evidence to suggest that while the 32 kbit/s ADPCM codec which complies with Recommendation G.726 exhibits the same distortion and additivity as PCM, the 16 kbit/s LD-CELP codec as tested and studied in 1991 exhibits additivity closer to 20 $\log_{10}(n)$. However, subjective tests carried out under the guidance of Study Group XII Experts Group on Speech Quality indicates that the 16 kbit/s codec pair closely tracks the subjective quality of the G.726 codec for up to 4 codecs in tandem. Beyond 4 in tandem, the 16 kbit/s codec performance decreases more rapidly than that of G.726 codec. Thus, it is proposed that the 16 kbit/s codec be treated the same as the G.726 codec determining network performance with the stipulation that no more than three 16 kbit/s codecs be allowed in the worldwide connection and noting that the LD-CELP distortion is not additive with the qdus of other codecs.

3.2 Sources of quantizing distortion

The units of quantizing distortion (qdu) tentatively assigned to a number of digital processes are given in Table 1. Background information on these assignments is given in Supplement Nos. 21 and 22, *Red Book*, Fascicles III.1 and III.2 (1985), respectively and in the notes associated with Table 1.

Conceptually the number of qdu assigned to a particular digital process should reflect the effect of only the quantization noise produced by the process on speech. In practice, the qdu must be determined from subjective measurements of real or simulated processes, where subjects will be exposed to not only the quantization noise but other impairments produced by the digital process tested.

Therefore, the subjective test results will be biased by these other impairments if the levels of these other impairments differ to a greater or lesser extent from the levels produced by PCM (the reference). Such biases will cause the derived qdu to not be a true measure of the effect of quantization distortion. The qdu assignment will instead reflect the effect of all the impairments on speech quality. Thus, to reduce the chance for such a bias to occur when determining the qdu assignments for digital processes, it is important to design the subjective test so as to:

1) minimize the contributions of impairments other than quantization distortion to the subjective test results; or

2) equalize the levels of these other impairments in the test and reference conditions.

TABLE 1/G.113

Planning values for quantizing distorsion

(Speech service only; see clause 4 for voice-band data considerations)
(see Notes 1, 11 and 12)

Digital process	Quantizing distortion units (qdu)	Notes
Processes involving A/D conversion		
8-bit PCM codec-pair (according to Recommendation G.711, A- or μ-law)	1	2, 3
7-bit PCM codec-pair (A- or μ-law)	3	3, 4, 5
Transmultiplexer pair based on 8-bit PCM, A- or μ-law (according to Recommendation G.792)	1	3
32 kbit/s ADPCM (with adaptive predictor) (combination of an 8-bit PCM codec pair and a PCM-ADPCM-PCM tandem conversion) (according to Recommendation G.726 or G.727)	3,5	6
16 kbit/s LD-CELP codec-pair (according to Recommendation G.728)	3,5	13
Purely digital processes		
Digital loss pad (8-bit PCM, A- or μ-law)	0,7	7
A/μ-law or μ/A-law converter (according to Recommendation G.711)	0,5	10
A/μ/A-law tandem conversion	0,5	
μ/A/μ-law tandem conversion	0,25	
PCM to ADPCM to PCM conversion (according to Recommendation G.721 or G.727)	2,5	8, 9
8-7-8 bit transcoding (A- or μ-law)	3	9

NOTES

1 As a general remark, the number of units of quantizing distortion entered for the different digital processes is that value which has been derived at a mean Gaussian signal level of about –20 dBm0. The cases dealt with in Supplement No. 21, *Red Book*, Fascicle III.1 are in accordance with this approach.

2 By definition.

3 For general planning purposes, half the value indicated may be assigned to either of the send or receive parts.

4 This system is not recommended by CCITT but is in use by some Administrations in their national networks.

5 The impairment indicated for this process is based on subjective tests.

6 Recommendations G.726 and G.727 perform equivalently at corresponding bit rates, including 24 and 40 kbit/s. However, qdu values cannot be assigned for 24 and 40 kbit/s operation, at this time.

7 The impairment indicated is about the same for all digital pad values in the range 1-8 dB. One exception is the 6 dB A-law pad which introduces negligible impairment for signals down to about –30 dBm0 and thus attracts 0 units for quantizing distorsion.

8 The value of 2.5 units was derived by subtracting the value for an 8-bit PCM codec pair from the 3.5 units determined subjectively for the combination of an 8-bit PCM code pair and a PCM/ADPCM/PCM conversion. Multiple synchronous digital conversions, such as PCM/ADPCM, PCM/ADPCM/PCM, are assigned a value of 2.5 units.

9 This process might be used in a digital speech interpolation system.

10 The qdu contribution made by coding law converters (e.g. μ-law to A-law) are assigned to the international part.

11 The qdu assignments to these digital processes reflect, to the extent possible, only the effect of quantization distortion on speech performance. Other impairments, such as circuit noise, echo and attenuation distorsion also affect speech performance. The effect of these other impairments must therefore be taken into account in the planning process.

12 The qdu impairments in this table are derived under the assumption of negligible bit error.

13 The distortion produced in the 16 kbit/s LD-CELP codec appears to be of a different nature than that denoted in terms of qdu in that it appears to add on a 20 $\log_{10}(n)$ basis. It is noted that one 16 kbit/s codec-pair produces a speech quality subjectively equivalent to that of one 32 kbit/s ADPCM codec-pair, while three 16 kbit/s codec-pairs produce a speech quality approximating that produced by four 32 kbit/s ADPCM codec-pairs. Thus, on the basis of this equivalence, one 16 kbit/s LD-CELP codec (according to Recommendation G.728) is assigned 3.5 qdu. It should be recognized that the qdu of the 16 kbit/s codec is not strictly additive with qdus of the other entries in the table.

3.3 Effect of random bit errors

The effect of random bit errors is under study.

3.4 Attenuation distortion and group-delay distortion

The provisional recommendation made in 3.1 specifies that the total quantizing distortion introduced by unintegrated digital processes in international telephone connections should be limited to a maximum of 14 units. It is expected that if this provisional recommendation is complied with, the accumulated attenuation distortion and the accumulated group-delay distortion introduced by unintegrated digital processes in such connections would also be kept within acceptable limits.

NOTE – The relationships among limitations imposed by quantizing distortion, attenuation distortion and group-delay distortion are under study.

3.5 Provisional planning rule

As a consequence of the relationship indicated in 3.4 above concerning quantizing distortion, attenuation distortion and group-delay distortion, it is possible to recommend a provisional planning rule governing the incorporation of unintegrated digital processes in international telephone connections. This provisional planning rule is in terms of units of transmission impairment which numerically are the same as the units of quantizing distortion allocated to specific digital processes as indicated in Table 1. The provisional planning rule is as follows:

The number of units of transmission impairment in an international telephone connection should not exceed: 5 + 4 + 5 = 14 units.

Under the above rule, each of the two national portions of an international telephone connection are permitted to introduce up to a maximum of 5 units of transmission impairment and the international portion up to a maximum of 4 units.

NOTE – It is recognized that in the mixed analogue/digital period, it might for a time not be practical for some countries to limit their national contributions to a maximum of 5 units of transmission impairment. To accommodate such countries, a temporary relaxation of the provisional planning rule is being permitted. Through this relaxation, the national portion of an international telephone connection would be permitted to introduce up to 7 units of transmission impairment. Theoretically, this could result in international telephone connections with a total of 18 qdu of transmission impairment. Such connections would introduce an additional transmission penalty insofar as voice telephone service is concerned. Administrations which find it indispensable to have a national allowance of more than 5 units (but no more than 7 units) should ensure that not more than a small percentage of traffic on national extensions exceeds 5 units.

3.6 Limitations of the provisional planning rule

In 3.5, it is assumed that for estimating the transmission impairment due to the presence of unintegrated digital processes in international telephone connections, the units of transmission impairment correspond to the units of quantizing distortion and that the simple addition of such units would apply.

For international telephone circuits that include tandem digital processes in an all-digital environment, adding the individual units of quantizing distortion might not accurately reflect the accumulated quantizing distortion (and, consequently, the accumulated units of transmission impairment). This could be the case since the individual amounts of quantizing distortion power produced by the individual digital processes might not be uncorrelated and, therefore, the addition of individual units of quantizing distortion might, under some circumstances, indicate totals that could be different from those actually in effect. This is explained in some detail in Supplement No. 21, *Red Book*, Fascicle III.1.

Although the 5 + 4 + 5 = 14 rule given in 3.5 might under some conditions provide only approximate results, the rule, nevertheless, is considered to be suitable for most planning purposes particularly in cases involving unintegrated digital processes. Examples of tandem digital processes which are explicitly taken into account in Table 1 are A-µ-A code conversion, µ-A-µ code conversion, and PCM-ADPCM-PCM conversion.

4 Effect of transmission impairments on voice-band data performance

Just as speech quality is affected by the transmission impairments found on telephone connections, so too is voice-band data quality. Many different impairments are present on a connection; some are steady-state impairments (e.g. loss, noise, quantization distortion, phase jitter, harmonic and intermodulation distortions, envelope delay distortion, echo, and attenuation distortion) while others are transient (e.g. impulse noise, phase or gain hits, and dropouts) and may tend to occur infrequently. Both steady-state and transient impairments can affect speech and voice-band data. However, the transient impairments almost always have a bigger impact on data than on speech. This is also true of some of the steady-state impairments, e.g. phase jitter and envelope delay distortion. Because of this, planning rules for circuits carrying speech usually concentrate on controlling the steady-state impairments, and less attention is paid to the transient impairments. If new planning rules are to be created with the intent of controlling the buildup of the impairments that are important to voice-band data, then these new rules will have to treat the transient as well as the steady-state impairments.

The extent to which certain impairments affect voice-band data depend upon the modem speed, modulation used and other characteristics such as whether the modem contains an equalizer to correct for envelope delay distortion. Low speed modems, operating at 1200 bit/s or less can usually tolerate a poorer SNR than higher speed modems. They also tend to be less sensitive to envelope delay distortion than the higher speed modems. Modems operating at 4800 bit/s and higher will usually contain an envelope delay distortion equalizer to minimize the effect of envelope delay distortion on the performance. Transients affect all modems, to a greater or lesser extent depending on many factors.

Two other factors influencing how impairments impact on voice-band data performance are:

a) whether error detection and/or correction techniques are employed; and

b) how the information to be sent is encoded.

If error correction is not used, then error causing impairments will cause errors in the output data. However, if error correction is used, then the impact of error causing impairments will only reduce the data throughput rate. Depending on how customer information is coded, errors can have more or less serious effects. For example, the loss of a letter in a word, because of a bit error in the 8 bits representing the letters of the alphabet, is probably less important than an error in the 8 bits used to convey information about the size, shape or location of a graphical symbol in an image.

Bit compression techniques such as ADPCM (according to Recommendation G.726) have a very significant effect on high speed ($\geq$ 4800 bit/s) modem performance.

Annex C gives results of studies using 32 kbit/s ADPCM. These results demonstrate the need for using higher encoding rates, such as 40 kbit/s ADPCM, for 9.6 kbit/s voice-band data (VBD) transmission. Modern digital circuit multiplication equipment (DCME, see Recommendation G.763) uses signal classifiers to detect higher speed VBD and encode it using 40 kbit/s ADPCM.

From the point of view of developing a simple planning rule which can be used to assess the effects of digital processes on voice-band data performance, several points are important:

1) Impairments (especially transients) other than those customarily measured for speech performance are important for measuring voice-band data performance.

2) A simple meaure of the steady-state impairments (e.g. signal-to-total noise ratio) may not prove to be a satisfactory basis for a voice-band data planning rule. A planning rule may have to take the transient impairments into account.

3) Modem type and speed must be taken into account. Thus, unlike the planning rules for speech, rules for voice-band data may turn out to be modem-specific.

4) The type of data service may influence the extent to which certain kinds of data errors and, thus, certain impairments are important. Therefore the planning rules may be service-specific.

5) Only an objective measurement method taking these first four points into account is likely to provide a successful basis for deriving useful planning rules.

6) Such a measurement method does not exist at present.

Therefore, until much more progress has been made in determining what impairments affect voice-band data performance, how to measure these impairments, what levels of these impairments are important, and how the differences in modem type, speed and other characteristics can be accounted for, this Recommendation must be limited in its application to speech services only.

Annex A

Information for planning purposes concerning attenuation distortion and group-delay distortion introduced by circuits and exchanges in the switched telephone network

(This annex forms an integral part of this Recommendation)

A.1 The information given in Tables A.1 to A.6 is derived from measurements[1] on modern equipment. The performance of actual connections in the switched telephone network can be expected to be worse than would be calculated from the tabulated data because of:

– mismatch and reflexion;
– unloaded subscribers' lines;
– loaded trunk-junctions with a low cutoff frequency;
– older equipment.

TABLE A.1/G.113

Two-wire local and primary exchanges

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	1.69	1.20	0.56	0.07
300	0.63	0.81	0.28	0.05
400	0.30	0.43	0.23	0.05
600	0	0.28	0.11	0.03
800	0	0	0.05	0.02
1000	–0.05	0.11	0.03	0.01
2000	–0.04	0.35	0	0
2400	–0.29	0.45	0	0
2800	–0.45	0.50	0	0
3000	–0.24	0.65	0	0
3400	–0.29	0.63	0	0
NOTE – The group-delay distortion may be taken to be with respect to about 2000 Hz.				

[1] Supplied by AT&T, Telecom Australia, Italy, British Telecom, NTT and Switzerland.

TABLE A.2/G.113

Four-wire exchanges

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	0.32	0.14	0.40	0.02
300	0.16	0.28	0.14	0.02
400	0.13	0.21	0.14	0.03
600	0.02	0	0.07	0.02
800	0	0	0.03	0.01
1000	0	0	0.02	0.01
2000	0.01	0.14	0	0
2400	0.06	0.21	0	0
2800	0.02	0.02	0	0
3000	0.10	0.07	0	0
3400	0.20	0.50	0	0

NOTE – The group-delay distortion may be taken to be with respect to about 2000 Hz.

TABLE A.3/G.113

Trunk junctions

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	4.29	1.95	3.05	0.36
300	0.86	0.49	1.42	0.18
400	0.36	0.31	0.78	0.09
600	0.09	0.17	0.34	0.06
800	0	0.03	0.16	0.02
1000	–0.03	0.04	0.08	0.02
2000	0.14	0.20	0.02	0.01
2400	0.33	0.29	0.06	0.03
2800	0.58	0.35	0.18	0.06
3000	0.88	0.55	0.31	0.11
3400	2.21	1.06	0.92	0.26

NOTES

1 The group-delay distortion may be taken to be with respect to about 1500 Hz.

2 The sample of trunk junctions included those on metallic lines, FDM and PCM systems.

3 PCM circuits may exhibit a somewhat lower attenuation distortion at 2000 Hz than that indicated above.

4 The values for trunk junctions are inclusive of 2-wire/4-wire terminations.

TABLE A.4/G.113

Circuits provided on a direct 12-channel group

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	1.56	0.92	5.42	0.22
300	0.39	0.43	2.97	0.35
400	0.11	0.30	1.45	0.22
600	0.05	0.18	0.76	0.10
800	0	0	0.44	0.05
1000	–0.01	0.11	0.26	0.02
2000	–0.03	0.19	0.01	0.01
2400	0.04	0.21	0.06	0.02
2800	0.13	0.33	0.21	0.04
3000	0.16	0.43	0.45	0.04
3400	1.03	0.56	1.97	0.20

NOTES

1 The group-delay distortion may be taken to be with respect to about 1800 Hz.

2 The data relates to 4 kHz channel translating equipment, the principal source of distortion in telephone circuits provided on direct 12-channel groups, i.e. circuits with only one circuit-section.

TABLE A.5/G.113

Circuits provided on a direct 16-channel group

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	2.80	1.63	9.74	0.40
300	0.04	0.19	4.39	0.27
400	–0.07	0.20	2.49	0.09
600	0.02	0.09	1.02	0.56
800	0	0	0.47	0.35
1000	0.09	0.08	0.19	0.28
2000	0.06	0.12	0.03	0.14
2400	0.03	0.14	0.36	0.31
2800	0.03	0.16	1.59	1.06
3000	–0.01	0.28	4.29	0.38

NOTES

1 The group-delay distortion may be taken to be with respect to about 1200 Hz.

2 The data relates to 3-kHz FDM channel translating equipment, the principal source of distortion in telephone circuits provided on direct 16-channel groups, i.e. circuits with only one circuit-section.

TABLE A.6/G.113

Circuits comprising three circuit-sections (4 kHz + 3 kHz + 4 kHz)

Frequency (Hz)	Attenuation distortion		Group-delay distortion	
	Mean value	Standard deviation	Mean value	Standard deviation
	(dB)	(dB)	(ms)	(ms)
200	5.92	2.09	20.58	0.51
300	0.82	0.64	10.33	0.56
400	0.15	0.47	5.39	0.32
600	0.12	0.27	2.54	0.58
800	0	0	1.35	0.36
1000	0.07	0.17	0.71	0.28
2000	0	0.29	0.05	0.14
2400	0.11	0.33	0.48	0.31
2800	0.29	0.49	2.01	1.06
3000	0.31	0.67	5.19	0.38

NOTES

1 This table has been derived from Tables A.4 and A.5, and relates to international circuits in which the middle section is routed on 3-kHz spaced channel equipment, e.g. a submarine circuit-section.

2 The group-delay distortion may be taken to be with respect to about 1400 Hz.

A.2 The reference frequency for attenuation distortion is 800 Hz. The reference frequency for group-delay distortion (i.e. the frequency at which the group delay is a minimum) has been estimated in each case.

A.3 In the results for circuits no allowance has been made for line signalling terminations although in some cases these distortions are included in the data for exchanges.

Annex B

Effect of transmission impairments on voice-band data

(From AT&T)

(This annex forms an integral part of this Recommendation)

B.1 Introduction

The present transmission plan for international connections provides guidance for the control of transmission performance, primarily to permit satisfactory transmission of speech signals. The significant impairments and their effect on speech signals are described in Recommendation P.11. These impairments include loudness loss, circuit noise, sidetone loudness loss, room noise, attenuation distortion, talker echo, listener echo, quantizing distortion and phase jitter. Other Recommendations involving data performance on leased circuits include H.12, M.1020 and M.1025.

The use of international connections for the transmission of non-speech signals such as voice-band data creates the need for increasing the scope of the transmission plan to include guidance on the control of additional impairments. The significant impairments for voice-band data include impulse noise, envelope delay distortion, phase jitter, non-linear distortion, tone-to-noise ratio, frequency shift, gain transients and phase transients. The following subclauses provide information on these impairments based on AT&T's experience. All the parameter values quoted are illustrative minimum end-to-end performance objectives of the *pre-divested* AT&T public switched network. Typical values obtained on the network are much better than the minimum objectives. These minimum values are considered to be consistent with satisfactory modem performance at speeds up to 4.8 kbit/s. More stringent minimum objectives are considered necessary for satisfactory performance at higher speeds such as 9.6 kbit/s. The parameter values shown are for illustration only and do not represent a proposed Recommendation.

B.2 Impulse noise

Impulse noise is defined as any excursion of the noise waveform on a channel which exceeds a specified level threshold. Impulse noise is evaluated on channels by counting the number of excursions during a predetermined time interval. In order to minimize contributions due to thermal noise, the minimum threshold is normally set 12 to 18 dB above the r.m.s. value of the noise. The impulse noise level is designated to be that threshold at which the average counting rate is equal to one per minute.

The measuring instruments used to count noise impulses may employ either electromechanical or electronic counters. In some sets, the maximum counting rate is controlled to be seven per second.

The contribution of impulse noise to error rate becomes significant when the noise peaks reach a level 3 to 12 dB below the r.m.s. data signal level depending upon: the type of modulation used by the data modems, the speed of transmission in bits per second, and the magnitudes of other transmission impairments on the channel. The minimum impulse noise objective is that no more than 15 counts in 15 minutes are to be tallied at a level above threshold which is 6 dB below the received data level. Control is exercised through engineering rules and limits on measured impulse noise levels.

Since most impulse noise originates as transients from the operation of relays and other switching equipment, engineering rules and mitigative measures are aimed at shielding low-level carrier signals from the radiation associated with these transients.

B.3 Envelope delay (group delay)

Envelope delay is defined as the derivative with respect to frequency of the phase characteristic of the channel. Measuring this derivative is impractical, so it is approximated by a difference measurement. There are numerous envelope delay measuring sets in use employing various frequency widths for this difference measurement. The AT&T standard is 166-2/3 Hz. In test results, these differences show up as varying resolution of ripples in the envelope delay characteristic. Narrow frequency widths yield higher resolution but reduced accuracy.

The frequency of minimum envelope delay in telecommunication channels is usually in the vicinity of 1800 Hz. Therefore, envelope delay measurements are usually normalized to zero at 1800 Hz. Departure from zero at other frequencies is referred to as envelope delay distortion. Envelope delay distortion gives rise to intersymbol interference in data transmission which causes errors and increased sensitivity to background noise.

In the network, envelope delay is controlled primarily in the design of channel bank filters and other apparatus. Typical minimum objectives for envelope delay distortion are 800 µsec maximum in the band from 1004 to 2404 Hz and 2600 µsec maximum in the band 604 to 2804 Hz.

B.4 Phase jitter

Phase jitter is defined as unwanted angular modulation of a transmitted signal. Its most commonly observed property is that it perturbs the zero crossings of a signal. Since noise also perturbs the zero crossings of a signal, it usually causes readings on a phase jitter measuring set even though no incidental modulation may be present.

Phase jitter impairs data transmission by reducing data receiver margin to other impairments. Phase jitter is controlled by the design of transmission equipment. Although specific sources of phase jitter, such as primary carrier frequency supplies, have been located in the field, the corrective techniques have usually required design changes in specific equipment. The end-to-end minimum objective for phase jitter is 10 degress peak-to-peak for the frequency band of 20 to 300 Hz and 15 degrees peak-to-peak for the band of 4 to 300 Hz.

B.5 Non-linear distortion

Non-linear elements in transmission equipment give rise to harmonic and intermodulation distortion which are more generally referred to as non-linear distortion. Non-linear distortion measurements are made usually in terms of intermodulation distortion measurements.

Non-linear distortion can be broadly defined as the generation of signal components from the transmitted signal that add to the transmitted signal usually in an undesired manner. The non-linear distortion of concern here is that found within an individual voice channel. It should not be confused with the intermodulation noise caused by non-linearities in the multiplex equipment and line amplifiers of a frequency division multiplex system. Although these non-linearities can contribute to the non-linear distortion at voice frequencies, their contribution is usually negligible.

Non-linear distortion is commonly measured and identified by the effect it has on certain signals. For example, if the signal is a tone having frequency A, the non-linear distortion appears as harmonics of the input, i.e. it appears as tones at $2A$, $3A$, etc. Since most of the distortion product energy usually occurs as the second and third harmonics, distortion is often quantified by measuring the power of each of these harmonics and is called second and third harmonic distortion. If the amount of non-linear distortion is measured by the power sum of all the harmonics, the result is called total harmonic distortion. These distortion powers are not meaningful unless the power of the wanted signal (the fundamental) is known, so measurements are usually referred to the power of the fundamental and termed second, third, or total harmonic distortion.

Historically, two different methods of measuring non-linear distortion on voice-band channels have been used: the signal-tone method and the 4-tone method. However, the single-tone method is no longer used.

For the 4-tone method, four equal level tones are transmitted as two sets of tones at a composite signal power of data level (–13 dBm0). One set consists of tones at 856 and 863 Hz (a 7-Hz spacing). A second set uses frequencies of 1374 and 1385 Hz (an 11-Hz spacing). The frequency spacing within each set of tones is not critical but should be different for each set. Let these four tones be called A_1, A_2, B_1 and B_2. The second order products ($A + B$) fall at $A_1 + B_1$, $A_1 + B_2$, $A_2 + B_1$ and $A_2 + B_2$. If the spacing between A_1 and A_2 is the same as that between B_1 and B_2 then $A_1 + B_2 = A_2 + B_1$ and these two components will add on a voltage basis and give an erroneous reading.

The third order products ($2B - A$) fall at $2B_1 - A_1$, $2B_1 - A_2$, $2B_2 - A_1$, $2B_2 - A_2$, $B_1 + B_2 - A_1$ and $B_1 + B_2 - A_2$. The receiver uses 50-Hz wide filters to select the $A + B$, $B - A$, and $2B - A$ products. R_2 is the ratio of the power of the received composite fundamentals to the power average of the $A + B$ and $B - A$ products. R_3 is the ratio of received composite fundamentals to the $2B - A$ products.

An advantage of the 4-tone method, the method currently used in AT&T, is that the 4-tone test signal has an amplitude density function quite similar to that of a data signal. However, because of the relatively wide (50 Hz) passband of the receiver filters, the measurements with the 4-tone method are more affected by circuit noise.

The intermodulation products arising from non-linear distortion add to the wanted signal and interfere with it much as noise does. The intermodulation products are more damaging than noise, however, and the ratio of fundamental to second- or third-order products should be in the range of 25 to 38 dB, depending upon the type of data transmission, for satisfactory operation.

Non-linear distortion is controlled primarily in the design of equipment. However, such things as aging vacuum tubes in older equipment and poor alignment of PCM channel banks can cause this distortion to increase over its design limits. The overall customer-to-customer minimum objective for non-linear distortion using the 4-tone method of measurement is 27 dB minimum for R_2 and 32 dB minimum for R_3.

B.6 Tone-to-noise ratio

For voice transmission, the noise that is heard during the quiet intervals of speech is most important and this is what the standard message circuit noise measurement evaluates. For data transmission, the noise on the channel during active transmission and corresponding signal-to-noise ratio is important. In systems using compandors or quantizers, the noise increases during active transmission. In order to measure this noise, a –16, –13, or –10 dBm0 tone is transmitted from the far end of the channel under test and then filtered out ahead of the noise measuring set. The filter used to remove the tone is a narrow notch filter centered at the frequency of the tone. This type of measurement is also referred to as noise-with-tone. Test equipment is now available which uses 1004 Hz as the tone for this measurement.

Noise, of course, can cause errors in data transmission and a tone signal-to-noise ratio objective of at least 24 dB should be maintained for satisfactory performance. Noise is controlled in the design of transmission equipment, in the engineering of transmission systems (by such factors as repeater spacing), and in the maintenance of these systems.

B.7 Frequency shift

When a tone experiences a change in frequency as it is transmitted over a channel, the channel is said to have frequency shift or offset. Frequency shift can be measured by using frequency counters at both ends of a channel. When the input frequency differs from the output frequency, the difference is the frequency shift on the channel.

In modem telecommunication equipment, the frequency shift, if any at all, is usually on the order of 1 Hz or less. Some older carrier systems may have substantial amounts of offset, e.g. 15 to 20 Hz.

Frequency shift is important in systems which use narrowband receiving filters such as telegraph multiplexers and remote meter reading equipment. When systems using these types of transmission experience frequency shift, the received signals fall outside the bandwidth of the filters. Frequency shift can occur on facilities which use single sideband suppressed carrier transmission. Within AT&T, frequency shift is controlled by means of the frequency synchronization network. The minimum objective for frequency shift is ± 5 Hz.

B.8 Gain and phase transients

Gain and phase changes that occur very rapidly may be encountered on telecommunication channels. Some of the more common causes of these phenomena are automatic switching to standby facilities or carrier supplies, patching out working facilities to perform routine maintenance, fades or path changes in microwave facilities, and noise transients coupled into carrier frequency sources. The channel gain and phase (or frequency) shift may return to its original value in a short time or remain at the new values indefinitely.

Gain changes are typically detected by changes in an automatic gain control circuit and phase changes by means of a phase locked loop. In order to provide protection against the test set detectors falsely operating on peaks of uncorrelated noise (impulse noise), a guard interval of 4 ms is designed into the gain or phase peak indicating instrument. Unfortunately, such a guard interval will also effectively make out true phase hits shorter than 4 ms that are not also accompanied by a peak amplitude excursion. The risk is considered justified at this time when one compares the known relative frequencies of occurrence of phase jumps to those of impulse noise.

Instrument used to measure gain and phase hits, as the rapid gain and phase changes are usually called, do so by monitoring the magnitude and phase of a sinusoidal tone. Hits are recorded and accumulated on counters with adjustable threshold levels. Gain hit counters typically accumulate events exceeding thresholds of 2, 3, 4 and 6 dB although they do not distinguish an increase from a decrease of magnitude. Similarly, phase hit counters accumulate changes at thresholds from 5 to 45 degrees in 5-degree steps. They respond to any hits equal to or in excess of the selected threshold. A switch which removes the impulse noise blanking feature under the user's discretion may be desirable when impulse phase hit activity is suspected. The wide variety in hit waveforms, the effect of noise on measurements, and the allowable tolerances in thresholds and measurement circuitry, will generally contribute to different hit counts even on instruments of identical design. This variability will lead to some confusing among those testing with hit counters of different manufacturers. An alternative specification of the entire hit counting circuitry is under further investigation by the Institute of Electrical and Electronic Engineers.

Gain hits begin to cause errors in high-speed data transmission when their magnitude is on the order of 2 to 3 dB. Phase hits begin to cause errors when their magnitude is about 20 to 25 degrees. The end-to-end minimum objective for gain hits is to have no more than eight gain hits exceeding 3 dB in 15 minutes; the minimum objective for phase hits is to have no more than eight phase hits in 15 minutes at a threshold of 20 degrees. A dropout is defined as a decrease in level greater than or equal to 12 dB lasting at least 4 ms. The minimum objective for dropouts is to have no more than two dropouts per hour.

Annex C

Adaptive differential pulse code modulation (ADPCM) performance impact on voice-band data

(From AT&T)

(This annex forms an integral part of this Recommendation)

Abstract

This annex is mainly based on an AT&T Bell Laboratories paper given at the "IEEE Global Telecommunications Conference" 2-5 December, 1985. It is provided to support this Recommendation as applied to voice-band data performance. The results indicate that, assigning a data qdu value to equipment using 32 kbit/s ADPCM (see Recommendation G.726) would be a difficult task since the performance is strongly dependent on the modem speed and type.

The annex reports on the results of a collection of empirical tests of high speed voice-band data modem error performance through channels containing asynchronously tandemed 32 kbit/s ADPCM (see Recommendation G.726) systems interspersed with simulated analogue impairments. A representative sample of 4.8 kbit/s transmision, and two 9.6 kbit/s devices were tested: an experimental design of Recommendation V.32 operating at 9.6 kbit/s for a full duplex modem, and another currently available 9.6 kbit/s product (similar to a V.29 modem). The results of the testing indicate that 4.8 kbit/s voice-band data transmission will perform adequately through asynchronous tandemed ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same conditions.

C.1 Introduction

It is possible to use adaptive differential pulse code modulation (ADPCM) at bit rates lower than 64 kbit/s per channel with, in many cases, less than proportional decrease in analogue transmission performance. Therefore, the use of a 32 kbit/s ADPCM algorithm on voice grade channels would essentially double the channel capacity of the associated facilities.

With the potential economic benefit due to increased capacity also comes the expectation of ensuing degradation of individual channel performance. Our results show that high speed voice-band data (e.g. 4.8 kbit/s or greater) would incur significant performance penalties with this new technology in place.

In this annex we report on the results of a collection of empirical tests of high speed voice-band data modem error performance through channels containing concatenated 32 kbit/s ADPCM (see Recommendation G.726) systems [1] interspersed with simulated analogue impairments. The channel configurations are designed to be representative of actual topologies possible on the public switched network with ADPCM systems in place. Asynchronously tandemed[2] ADPCM hardware contained in these test channels range in number from zero to seven while the interspersed analogue impairments are obtained by allocating parameters from impairment distributions measured in the end office connections study (EOCS) [2], loop studying 1970 [3], and 1980 loop surveys. We also tested performance using connections with asynchronously tandemed 64 kbit/s PCM systems, implemented in D4 channel banks, to compare with ADPCM configurations that showed particularly poor performance, so that it could be determined whether the ADPCM algorithm or simply the PCM coding was at root.

Modems used for the testing were of the high speed type. We tested a representative sample of 4.8 kbit/s transmission (V.29 type), and two 9.6 kbit/s modems: an experimental design of the V.32 modem standard for a full duplex modem, and another currently available device (V.29 type). All of these devices are 2-2 wire modems which are, or will be, marketed for use on the public switched network.

The results of our testing indicate that 4.8 kbit/s voice-band data transmission will perform adequately through multiple asynchronous tandeming of ADPCM systems, but that 9.6 kbit/s transmission is limited and, with certain modems, unacceptable under the same configurations.

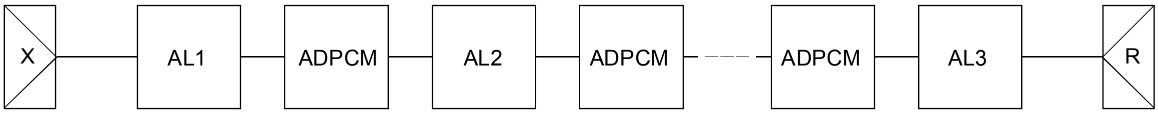
C.2 Test condition architecture

It is known that ADPCM algorithm precision is to a great extent dependent on the nature of the signal which is to be encoded and transmitted. Signals with little or no stochastic components, such as pure tones, traverse these systems very well, with little or no distortion. On the other hand, high speed voice-band data signals which inherently have a large stochastic component and substantial bandwidth are significantly affected by ADPCM coding. Due to this, our test condition architecture examines these high speed modem types. We have furthermore tried to efficiently limit the quantity of testing required by using a universal architecture template for all our studies.

C.2.1 4.8 kbit/s half-duplex

Figure C.1 shows the test configuration architecture for 4.8 kbit/s half-duplex testing. The configuration is shown terminated on both ends with modems. The sequence of additional apparatus on the chart begins from the left with simulated analogue impairments (AL1) representative of analogue loop and access trunk (AT). Then the long haul segment consists of an ADPCM system, one 500 mile equivalent L-carrier analogue link (AL2) followed by from 1 to 6 ADPCM's respectively. This structure is representative of an interexchange portion consisting of multiple links and models the segment as if all analogue impairments occur early in the segment. Although this placement of the analogue impairments is somewhat conservative, it is counterbalanced by the fact that the impairments are those of a single L-carrier link and is a good approximation of reality given the constraint of using a single impairment simulator for the long haul part. Finally egress to the receiver proceeds through another analogue impairment simulator (AL3) representative of analogue trunk and loop. Interspersing analogue impairments with ADPCMs in this manner for the connection is more representative of actual network topologies and applications than simply lumping all analogue impairments in one place.

[2] Asynchronous tandeming takes place when a previously ADPCM coded signal is decoded to its analogue version and then recoded in a subsequent ADPCM system.



Modem	Access	LH			Egress	Modem
	Loop&AT	ADPCM	1 LMX	n (ADPCM)	AT&Loop	# tests
4.8 kbit/s	85	None	85	None	85	1
	None	ADPCM	None	1-6 ADPCM	None	6
	None	PCM	None	1-6 PCM	None	6 AN
	µ, 85	ADPCM	µ, 85	1-6 ADPCM	µ, 85	48
	µ, 85	PCM	µ, 85	1-6 PCM	µ, 85	48 AN

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FIGURE C.1/G.113

Test condition architecture for 4.8 kbit/s modem

It is clearly necessary to determine, for this configuration, the type and actual values of the analogue impairments to be dialed into simulators AL1, AL2 and AL3. Using a network performance modeling tool, the results of the end office connections study (EOCS), and the assumption that high speed data customers connect to the network via data jacks, we derived the end-to-end mean (M) and 85th percent conditions of the major subset of impairments for switched network channels. Note that although we refer to the channel with each impairment at the 85 percent level as the 85th percentile channel, in fact it is somewhat worse because all impairments at 85% in one channel simultaneously would actually appear less than 15% of the time. Nevertheless, we then allocated these end-to-end values to the analogue impairment simulators. The results of this allocation, the impairment types, and the end-to-end values are shown in Table C.1. The values designated are allocated from the end-to-end mean (M), while the values designated "85" are allocated from the 85% end-to-end impairment values. The discussion of Figure C.1 can now be completed by describing the various values of analogue impairments as well as type and number of digital equipment present. The first configuration shows no ADPCMs but contains the allocated impairments from the 85th percent channel. Next, for additional reference, we tested six channels containing from 2 to 7 ADPCMs only, with no analogue impairments. Another six channels were to be tested as necessary with only PCM devices asynchronously tandemed, if and only if the previous corresponding ADPCM tests showed poor performance. Finally, the important tests with both analogue impairments allocated to the simulators from the mean (µ) and 85th percent channel with from 2 to 7 ADPCMs (or PCMs as necessary) were performed.

C.2.2 9.6 kbit/s full and half-duplex

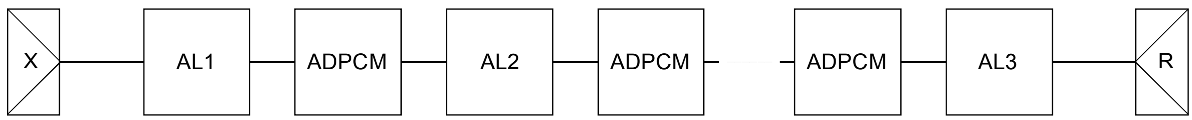
Here the test configuration architecture template is shown with a chart in Figure C.2. An experimental implementation of the V.32 modem standard for a 9.6 kbit/s full-duplex modem was tested under identical values of analogue impairments as those used for the 4.8 kbit/s modem. Although the channel segments have the same representation in the template, we only tested from 1 to 3 ADPCMs in the long haul segment. The simulated full-duplex operation was tested with the opposite channel excited with data, a signal-to-listener echo ratio of 12 dB, and a listener echo delay of 25 ms, in line with tests previously reported to Study Group XVIII [4]. For these tests, Table C.1 again has the relevant values for the analogue impairment simulators.

Also shown are three tests of another 9.6 kbit/s half-duplex modem with ADPCMs only. This modem is specifically designed for use on the public switched network and represents expected performance of the most currently available 9.6 kbit/s technology.

TABLE C.1/G.113

EOCS derived test conditions

	AL1	AL2	AL3	E-E
Impairment	μ/85	μ/85	μ/85	M/85
Loss (dB)	11.0/11.4	1.1/1.7	11.0/11.4	23.0/24.5
C-notch noise (dBmC)	32.0/35.6	37.5/38.5	24.0/27.6	29.4/31.0
Slope (dB)	1.5/3.0	0.0/0.2	1.5/3.0	2.9/6.1
Env. delay distortion (μs)	226/388	632/755	226/388	1084/1535
2nd intermod. (dB)	66.0/50.2	58.4/53.8	66.0/50.2	52.7/46.3
3rd intermod. (dB)	74.0/53.0	56.9/50.3	74.0/53.0	51.7/44.3
Phase jitter (p-p)	0.5/0.7	1.9/3.7	0.5/0.7	3.5/5.1
Level (dBm)				–27.0/28.5
S/N (dB)				31.6/28.5



Modem	Access	LH			Egress	Modem
	Loop&AT	ADPCM	1 LMX	n (ADPCM)	AT&Loop	# tests
V.32	85	None	85	None	85	1
9.6 kbit/s	None	ADPCM	None	1-3 ADPCM	None	3
V.32	None	PCM	None	1-3 PCM	None	3 AN
	μ, 85	ADPCM	μ, 85	1-3 ADPCM	μ, 85	24
	μ, 85	PCM	μ, 85	1-3 PCM	μ, 85	24 AN

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FIGURE C.2/G.113

Test condition architecture for 9.6 kbit/s modems

C.2.3 4.8 kbit/s ADPCM performance

For 4.8 kbit/s transmission, the salient results are shown in Figure C.3. We have plotted four curves on the axes: two 1000-bit block error rates (BLER) and two bit error rates (BER), one each for the mean and 85% EOCS channels. The abscissa counts the number of asynchronously tandemed ADPCMs in the connection. Due to the architecture of the tests these are enumerated as $1 + n$. The "1" represents the ADPCM between AL1 and AL2 while n is the number of ADPCM systems between AL2 and AL3.

We see clearly from the graphs that all the error performance measures degrade as the number of asynchronously tandemed ADPCMs increases, and that performance on the 85% channel, containing worse values of analogue impairments, is inferior to the mean channel results. We assume an acceptance limit for modem accuracy behaviour of a BER < 10^{-5} on 85% of channels and a BLER < 10^{-2} on 85% channels. Hence, if we focus on the 85% channel from EOCS, we see that 4.8 kbit/s performance will be at acceptable limits if the number of ADPCMs is between 4 and 5 for BLER and between 3 and 4 for BER. More recent results imply that for some modems the BER criteria is marginal with 3 in tandem and only 2 would be acceptable. We know of course that the BER criterion is stricter than the BLER limit because bit errors represent a greater burst phenomenon which is to a large extent ameliorated by the use of block transmission implemented with an error detection/correction protocol. Nevertheless, we tested and present both results because customer data communication applications will dictate which measure is more relevant.

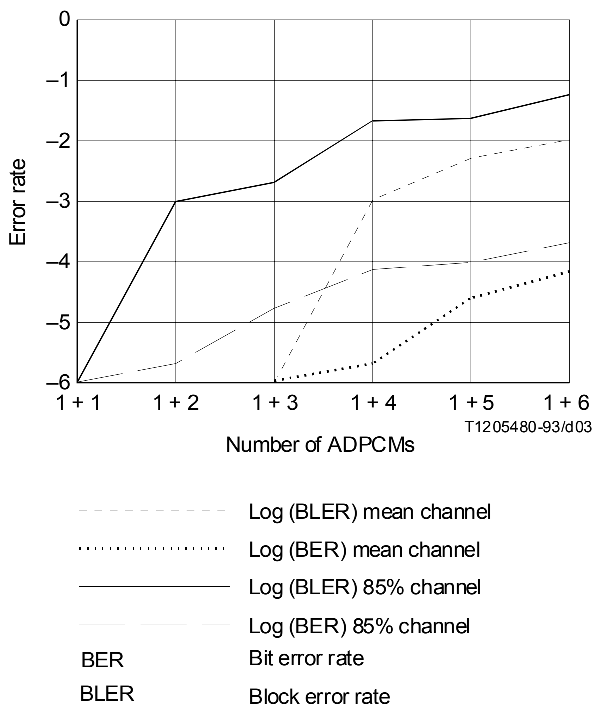

FIGURE C.3/G.113

ADPCM performance (mean and 85% channels) with a 4.8 kbit/s modem

C.2.4 V.32 modem-ADPCM performance

The outcome of tests on the experimental testbed representing a 9.6 kbit/s device conform to Recommendation V.32 is shown in Figure C.4. Note that we have again plotted four performance curves. As before, performance of the 85th percent channel is inferior to that of the mean channel. If we now focus on the 85th percent channel BLER, we see that the acceptable performance limit occurs between 2 and 3 asynchronously tandemed ADPCMs, while for BER the number is somewhere between 0 and 1. Which performance measure is appropriate depends on customer application. We are here observing that a larger stochastic component of the data signal implies poorer error performance of the modem. In this case the use of 9.6 kbit/s shows a definite degradation in performance over the same topology with 4.8 kbit/s devices.

It is also interesting to see if changing the position of segments with poorer impairment values effects modem performance. Figure C.5 shows a graph of three BLER curves for V.32 modems where we have taken the allocated 85th percent segment first on access, then on the long-haul part, and finally on the egress of the test channel, the other segments being at the allocated mean values of impairments. First, note that these curves fall between the full 85th percent channel and the mean channel in performance. Next, note that there does appear to be a mild dependence on the location of the more severe impairment values. Worse impairments close to the transmitter appear to have a more destructive effective on modem BLER performance than if they appear closer to the receiver. This means that analogue impairments on access are probably more significant in affecting modem error rates than those in the long-haul network or egress. The observed effect is mild, however, probably because the impairment values of the allocated 85th percent segments are really not much poorer than those for the allocated mean segments.

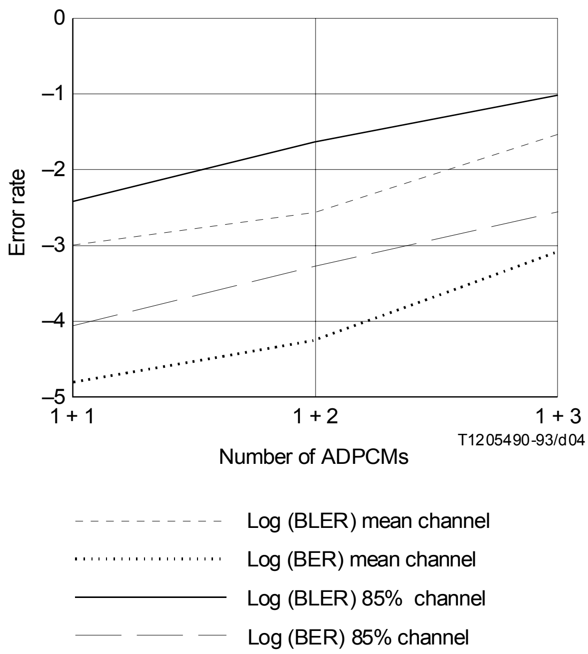

FIGURE C.4/G.113

ADPCM performance (mean and 85% channels) with a V.32 modem

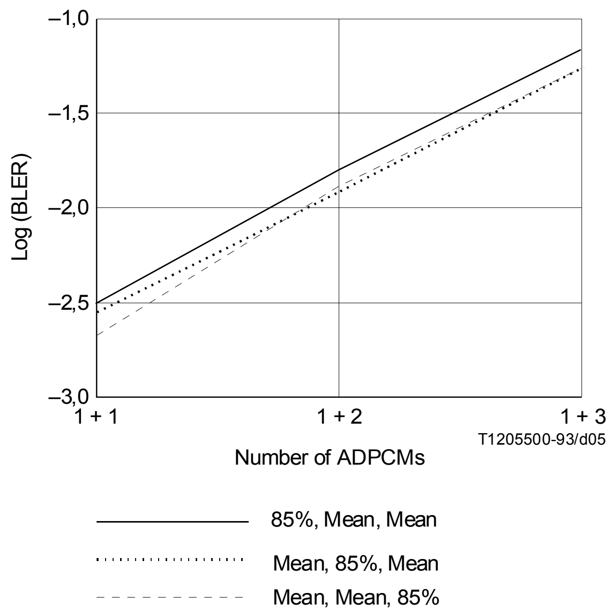

FIGURE C.5/G.113

ADPCM performance (impairment position study) with a V.32 modem

C.2.5 9.6 kbit/s – ADPCM performance

As a final test of modem performance, we have subjected another 9.6 kbit/s device, utilizing more traditional technology, to a sequence of asynchronously tandemed ADPCMs. This modem is a 2-wire device advertised by the vendor for use on the public switched network at signalling rates to 9.6 kbit/s. We have tested the device performance with no analogue impairments at all in the test channel. During the course of the empirical determination, it was discovered that the modem start sequence and the ADPCM algorithm interacted to prevent commencement of communication between transmitter and receiver. It was therefore necessary to test by allowing modem training to occur on an ordinary PCM channel after which ADPCMs were cut in to observe performance. Similar availability problems would also probably occur for any speed modem whose start-up training sequence is similar to that of this 9.6 kbit/s product.

Figure C.6 shows the performance results for this modem. Without analogue impairments the number of ADPCMs may simply be enumerated sequentially. The BLER outcome indicates that between 0 and 1 ADPCM encoding is all that can meet our performance criterion. For BER it appears, again by our normal criterion, that ADPCM is incompatible with proper operation of the modem. Since it is expected that many modem vendors will, or have already, announced high speed 2-wire devices for use on the public switched network, the presence of ADPCM on these channels is likely to cause performance problems for those devices which are similar to the one tested for training, modulation and detection.

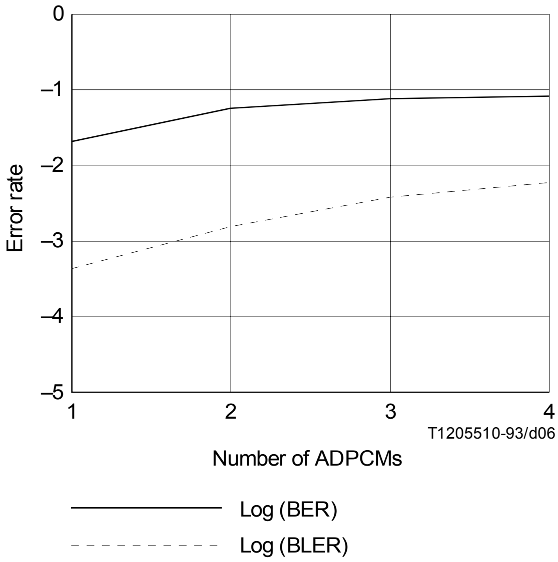

FIGURE C.6/G.113

9.6 kbit/s ADPCM performance (no analogue impairments)
with a 9.6 kbit/s modem

C.3 Conclusions

In this annex we have reported on the architecture, laboratory apparatus, and results of a collection of empirical tests of high speed voice-band data modem error performance through channels containing asynchronously tandemed ADPCM systems interspersed with simulated analogue impairments. The results are compactly displayed in Table C.2 which shows that communication at 4.8 kbit/s may proceed through more asynchronous tandemed ADPCMs than in the case of using 9.6 kbit/s devices. Furthermore, communication at 9.6 kbit/s can be unacceptable when a BER criterion is applied, but sometimes acceptable when a BLER criterion is applicable. Clearly the appropriate criterion depends on the data communication user's application.

TABLE C.2/G.113

Number of allowed ADPCMs on EOCS 85% channel

Modem	BER = 10^{-5}	BLER = 10^{-2}
4.8 kbit/s (V.29)	3/4[a)]	4/5
V.32	0/1	2/3
9.6 kbit/s	0	0/1

a) More recent results imply the range is 2/4.

References

[1] Draft-Proposed American National Standard *32 kbit/s ADPCM Algorithm and Line Format*, Committee T1, Subcommittee T1Y1, Document No. T1Y1, LB 85-01, 28 March, 1985.

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[3] MANHIRE (L. M.): Physical and Transmission Characteristics of Customer Loop Plant, *BSTJ*, 57, 1/70.

[4] KALB (M.), MORTON (C. H.) and SHYNK (JU. J.): DATACAL – A Voice-band Data Communication Connection Performance Model, *Proc. of the Second International Network Planning Symposium, University of Sussex*, Brighton, UK, 21-25 March, 1983.

Annex D

Compatibility of speech coding algorithms and voice-band data

(This annex forms an integral part of this Recommendation)

As described in Annexes B and C digital speech coders have an impact on voice-band data. The following list provides some approximate information concerning the capability of various speech coding algorithms to support voice-band data signals.

1) The 16 kbit/s LD-CELP (see Recommendation G.728) algorithm supports voice-band data only up to 2400 bit/s.

2) The 32 kbit/s ADPCM (see Recommendation G.726) supports voice-band data up to 4800 bit/s.

3) The 40 kbit/s ADPCM (see Recommendation G.726) supports voice-band data up to 9600 bit/s, with 14 400 bit/s supported for only non-tandem connections.

This list will be expanded in the future. The list is intended for general guidance only. Annex C provides more detail regarding item 2, and additional data regarding item 3 can be found in [1].

Reference

[1] DIMOLITSAS (S.), CORCORAN (F. L.), ONUFRY (M.) and SUYDERHOUD (H. G.): Evaluation of ADPCM Coders for Digital Circuit Multiplication Equipment, *COMSAT Technical Review*, Vol. 17, No. 2, pp. 323-345, Fall 1987.

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