International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

Y.1541

Amendment 2

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SERIES Y: GLOBAL INFORMATION INFRASTRUCTURE, INTERNET PROTOCOL ASPECTS AND NEXT-GENERATION NETWORKS

Internet protocol aspects – Quality of service and network performance

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Network performance objectives for IP-based services

Amendment 2: New Appendix XI – Digital circuit (ISDN) emulation requirements on IP-based networks

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ITU-T Recommendation Y.1541

Network performance objectives for IP-based services

Amendment 2

New Appendix XI – Digital circuit (ISDN) emulation requirements on IP-based networks

Source

Amendment 2 to ITU-T Recommendation Y.1541 (2006) was agreed on 25 January 2007 by ITU-T Study Group 12 (2005-2008).

FOREWORD

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

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

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

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(This appendix does not form an integral part of this Recommendation)

XI.1 Introduction

The purpose of this appendix is to derive a packet loss requirement for the support of the I.231.1 ISDN clearmode service over an IP network, based on the requirements of [ITU-T G.826] for error rates on transmission systems carrying ISDN connections. Next, the possible mitigation of the packet loss requirement through the use of forward error correction (FEC) techniques is examined.

We consider a multiplex of RTP packet streams as an emulation of a TDM digital transport connection, which should meet the G.826 requirements for digital connections. This analysis is based on the material presented in Delayed Contributions [ITU-T COM12-D17] and [ITU-T COM12-D83].

XI.2 Packetization and transport assumptions

Consider transmission of a single ISDN B-channel as a 64 kbit/s stream in 10 ms packets, 100 packets/s per stream. The resulting stream of RTP packets has a 9-octet POS overhead, 40-octet RTP/UDP/IP headers and 80-octet user data payload.

Suppose then that the packetized B-channel is routed with other packetized 64 kbit/s streams (voice and ISDN) across a core network using STM-1 transmission systems. This system transports 92.9 Mbit/s user data payload using an overall bit rate of 149.76 Mbit/s. Block size at STM-1 is 18'792 bits and there are 8'000 blocks/s. Thus a given 64 kbit/s stream contributes a packet to every 80th block and the multiplex can carry about 149'760'000/(129 * 8 * 100) = 1'451 64 kbit/s streams. A block contains about 18 packets, all from different 64 kbit/s streams.

XI.3 Range of packet loss requirements

This clause derives the UNI-UNI packet loss ratio necessary to meet various transport accuracy requirements.

The first approach to derive a packet loss requirement refers to the G.826 specification for a background block error ratio (BBER) of 2×10^{-4}, and loss of a single packet will result in a background block error. Hence packet loss ratio must be better than $2 \times 10^{-4}/18 = 1.1 \times 10^{-5}$ to meet the BBER specification.

The second approach examines the G.826 errored second ratio (ESR) of 0.16. Since loss of a single packet will result in an ES, and there are 145'100 packets/s, the packet loss ratio must be better than $0.16/145'100 = 1.1 \times 10^{-6}$ assuming random packet loss. Because 0.16 is not very much less than 1, there is a small correction arising from the finite probability of two errors in the same second, which we have neglected for this approximate calculation. The packet loss ratio derived from ESR is about 10 times more demanding than from the BBER derivation, so the more stringent requirement of these two will be adopted.

The third approach considers the G.826 specifications for ESR and severely errored second ratio (SESR) for sub-primary rate connections. With 10 ms packetization, loss of a packet results in loss of 640 payload bits which must be replaced by dummy data. On average 320 bits will be in error, and an SES is a second in which the error ratio is 1.0×10^{-3}, so an SES will result for connections at rates less than or equal to 320 kbit/s (5×64 kbit/s, requiring 500 packets/s). The SESR for sub-primary-rate connections is 2×10^{-3}. Hence the packet loss ratio must be less than $2 \times 10^{-3}/500 = 4 \times 10^{-6}$ assuming random packet loss.

Thus, the different requirements within [ITU-T G.826] lead to somewhat different values of packet loss for an international IP network. The derived requirements range between 1.1×10^{-6} and 4×10^{-6} depending on the assumptions above and the specification from [ITU-T G.826]. However, both these figures are very much more stringent than the IPLR of 1×10^{-3} for QoS Classes 0 through 4 of Y.1541.

XI.4 Effect of forward error correction

An alternative to achieving the very low packet loss ratio needed for the I.231.1 service is to use forward error correction, trading bandwidth and extra delay for a less-demanding packet loss requirement. [IETF RFC 2733] describes a scheme for protected transmission of RTP streams through networks with packet loss. The scheme permits the use of a range of block FEC methods.

For example, (n, k) block codes may be used, generating n-k redundant packets from each k data packets and transmitting all n packets. All k data packets can be recovered provided any k packets of the n are received without loss or error. The probability of a residual error (one not corrected by the scheme) affecting a block, P_b, is equal to the probability of losing more than n-k packets from the block, hence approximately equal to the probability of losing $n - k + 1$ packets from the block. If packet loss is random, this is given by:

$$P_b \approx \Pr(n-k+1) = \frac{n!}{(n-k+1)!(k-1)!} p^{n-k+1}(1-p)^{k-1}$$

where p is the probability of loss of a single packet. As $p \ll 1$, the term involving $(1 - p)$ will always be close to 1 for the expected small values of k.

Consider a stream of payload packets of rate R packets/s. The rate of generation of blocks is R/k. Hence the rate at which such blocks suffer from loss of more than n-k packets is RP_b/k. If a block suffers from loss of more than n-k packets, the worst case is that no payload packet is recoverable from the block, so the worst-case rate of loss of payload packets after FEC is $kRP_b/k=RP_b$. This is to be compared with a rate of loss of payload packets of $(R \times p)$ in the absence of FEC. Thus, P_b is an effective packet loss probability after FEC. P_b may be an overestimate of the effective packet loss probability, in cases where the FEC code allows recovery of some payload packets even after loss of more than n-k packets from the transmitted block.

To achieve the packet loss requirements derived above, we wish to make the post-FEC packet loss probability around 1×10^{-6} when operating on an IPLR of $p = 1 \times 10^{-3}$, which is assured on paths that are compliant with Y.1541 Classes 0 through 4.

A (k + 1, k) scheme was chosen for further analysis, because of its simplicity. Any (k + 1, k) block code which adds a single parity packet leads to a probability of a residual block error equal to a numerical factor (greater than 1) multiplied by p^2. The two simplest such codes are (2,1) (simple repetition, requiring double bandwidth) and (3,2) (needing only the exclusive-OR operation, increasing bandwidth by 3/2).

The probability of an error in the block for the (2,1) code is just p^2. For the (3,2) code, it is $3p^2$. For a loss requirement of 1.1×10^{-6} after correction, the requirement before correction is 1.05×10^{-3} for the (2,1) repetition "code". For the (3,2) code, the requirement before correction is 6.0×10^{-4}. Note that these are requirements on total packet loss, including both IPLR and packets arriving too late to be played out.

The 1.05×10^{-3} requirement is numerically close to the IPLR values for QoS Classes 0 and 1, however this does not account for additional packets that arrive too late for play-out as permitted by the $1 - 10^{-3}$ quantile used in the IPDV specification.

Provisional QoS Classes 5 and 6 offer much more stringent IPLR objectives and IPDV quantiles (1×10^{-5} and $1 - 10^{-5}$, respectively). Using the overall loss based on these values, it is clear that an FEC code can be designed to meet the loss requirement of 1.1×10^{-6} after correction using much less overhead. For example, a (14,13) code can correct a $p = 10^{-4}$ loss ratio to $91p^2$, or 9.1×10^{-7}.

The desire for low overhead must be tempered by the delay consumed in the FEC processing. Delay increases by at least $(k - 1)$ times the packetization time.

XI.5 References

[ITU-T COM12-D17] ITU-T Delayed Contribution COM12-D17 (2005), *IP packet loss and support for emulated I.231.1 ISDN "clearmode" service*.

[ITU-T COM12-D83] ITU-T Delayed Contribution COM12-D83 (2005), *On the overall packet loss specification for ISDN emulation*.

[IETF RFC 2733] IETF RFC 2733 (1999), *An RTP Payload Format for Generic Forward Error Correction*.

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