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Methodology for deriving performance objectives and its optimization for IP packet applications in the mobile-satellite service

M Series

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International Telecommunication Union

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Note: *This ITU-R Recommendation was approved in English under the procedure detailed in Resolution ITU-R 1.*

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RECOMMENDATION ITU-R M.1741*

Methodology for deriving performance objectives and its optimization for IP packet applications in the mobile-satellite service

(Questions ITU-R 85/8, ITU-R 87/8, ITU-R 112/8 and ITU-R 233/8)

(2006)

Scope

This Recommendation stipulates the methodology for deriving performance objectives and its optimization for IP packet applications in the mobile-satellite service. The guidelines for the performance parameters and objectives for physical and MAC layers, the methodology for derivation of the performance objectives, and the guidelines for the optimization of TCP performance in IP packet data applications in the mobile-satellite service are provided in Annexes 1, 2 and 3 to this Recommendation, respectively.

The ITU Radiocommunication Assembly,

considering

a) that Internet Protocol (IP) packet transmission has become one of the major applications in modern communication networks including mobile-satellite systems;

b) that hypothetical reference circuits, technical characteristics, performance objectives and availability requirements have been stipulated for conventional mobile-satellite services (MSSs) in a number of existing Recommendations;

c) that technical characteristics and performance should be defined on the basis of IP packet layers, in addition to basic digital transmission performance of the MSS bearer link;

d) that Recommendation ITU-R M.1636 stipulates definitions for reference models and performance parameters as a technical basis for the development of IP packet applications in the MSS;

e) that the guideline for the performance objectives of the physical and link layers of MSS systems serving IP packets, together with the methodology to derive performance objectives for IP packet transmission for the system are required to facilitate effective use of spectrum resources by the system;

f) that the transmission control protocol (TCP) is one of the most widely spread transport layer protocols over IP, where special attention is required for optimization of the operational parameters if it is applied to a system with large transmission delay such as MSS links,

recommends

1 that the guidelines for performance objectives in Annex 1 should be used for IP packet transmission in the MSS;

2 that the methodology in Annex 2 should be applied for derivation of the performance objective for IP packet applications in the MSS;

3 that the guidelines in Annex 3 should be taken into account when the operational parameters of TCP over IP packet transmission in the MSS are determined.

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* This Recommendation should be brought to the attention of Radiocommunication Study Group 4 and Telecommunication Standardization Study Group 13.

Annex 1

Guidelines for performance objectives for IP packet transfer in the MSS

1 Introduction

Terminals in the MSS have a number of performance requirements regarding employed protocols to ensure that the characteristics of the satellite link (such as large delay, variable error rate, a periodic disruption) are tolerated and adequately handled. The protocols to be considered, which operate under IP packet transmission over the satellite link, include the physical (PHY) and the media access control (MAC) layers. This section identifies the characteristics and performance of PHY and MAC layers that can contribute to the performance of IP packet transmission over the MSS link.

2 Physical layer (Layer 1)

2.1 Packet-based multiple access channel

Radio channels and their resources are generally accessed by a number of mobile-satellite terminals to connect to satellite access gateways through an MSS link for IP packet transmission. The sharing mechanisms of the radio channels and their resources for the packet-based multiple access involve a combination of techniques. One of the possible resource management approaches for the mobile-satellite terminals to access the radio channels is to utilize the channels in the forward direction on a time-division-multiplex (TDM) basis, and in the return direction on a time-division-multiple-access (TDMA) basis. In this case, each satellite access gateway manages a set of forward (to-mobile) and return (from-mobile) channels. In addition, mobile-satellite terminals may support one or more transmit/receiver pairs.

The TDMA for the physical layer requires certain transmission scheduling protocols, which are described in detail in § 4.4.

2.2 Physical layer roles

The PHY layer is responsible for the transfer of an information bit stream over the satellite link, and includes the following functions:

At a transmitter:

– encoding, scrambling and interleaving;

– modulation of the encoded bit stream;

– transmission of the modulated signal on a multiplexed channel.

At a receiver:

– reception of a modulated signal;

– demodulation of the signal into an encoded bit stream;

– decoding, descrambling and de-interleaving.

The PHY layer transfers an entire frame together with the measured information concerning parameters of the satellite signal such as timing and power level.

3 MAC layer (Layer 2)

The MAC layer is responsible for controlling access to the PHY layer (channel resources) by each connection and for the establishment and provision of the connection for the delivery of IP packets over the MSS link. The MAC layer generally performs the following functions:

– addressing of physical devices or logical connections, and scheduling of resources for transfer of information between these entities in both the forward and return directions on a packet-based multiplexed channel;

– packing and unpacking of IP packets into MAC frames, including segmentation and reassembly, as required;

– buffering and flow control of information from the upper layer (i.e. IP layer);

– automatic repeat request (ARQ) (if required for the particular, error sensitive connection).

4 Guideline for designing IP packet data transmission system in the MSS which may influence performance of the network layer (Layer 3) and higher

This subsection describes the performance parameters and guidelines for system design of PHY and MAC layers for IP packet transmission in MSS, which can affect the performance of IP packet transmission over the MSS link.

4.1 Bit error ratio

The bit error ratio (BER) for the MSS link can be derived from the design of the MSS link that depends on the physical layer characteristics of the link such as the modulation and the error coding scheme, transmitting power and link margin, receiver sensitivity and so forth. An example of BER performance for a 64 kbit/s bearer channel is shown in Table 1.

TABLE 1

Example of BER performance

Channel	MSS terminal demodulator BER	MSS gateway demodulator BER	Measurement period
64 kbit/s	$\leqq 10^{-6}$	$\leqq 10^{-6}$	1 500 s

The values of the example in Table 1 are measured after synchronization of the signal has been established. All possible causes of the BER degradation, such as phase noise, loss of synchronization lock, clock and cycle slips during the measurement are included but any degradation caused by burst error due to multipath fading, blockage, shadowing or adjacent channel interference is excluded.

4.2 Buffering and flow control

The IP packets arriving from the upper layer (i.e. IP networking layer) will be buffered at the MAC connection sub-layer before being transmitted by the packet-based multiple access channel of the PHY layer. The control mechanism and the size of the buffer will affect the performance of the IP packet transmission. For example, if the buffer size is sufficiently large, loss of packets due to the buffer overflow can be decreased, while the buffering delay can increase significantly if the buffer size is too large. Therefore, the mechanism of the buffer control, including the size of the buffer, needs to be determined carefully taking account of the trade-off between the loss and the delay of IP packet transmission.

4.3 ARQ

The MAC layer offers a facility for reliable transmission, whereby information is delivered error-free and in-sequence, at the expense of an increase in packet transmission delay. The mechanism for support of ARQ also needs to be selected such that the trade-off between error and delay performance can be well balanced.

4.4 Scheduling

Scheduling is the process to allocate resources of the packet-based multiple access channel to each connection, which is managed by the satellite gateway. The simple method to allocate resources is first-in first-out (FIFO), in which priorities of all connections are considered as equal. In order to differentiate the priority of each connection, the scheduling process may be based on the quality of the service parameters of each connection.

In addition, the status of the queues for other connections might be used to determine the priority to transfer data in the scheduling process.

Annex 2

Methodology for deriving IP packet transfer performance parameters in the MSS

1 Introduction

The generic definitions of the performance parameters and their objectives are provided in ITU-T Recommendations Y.1540 and Y.1541, respectively, for IP packet-based applications. For IP packet transmission using the MSS, dominant components for such performance parameters are contributed by the MSS section of the end-to-end communication path. Taking account of the properties and the performance parameters of the MSS links defined in the previous section, this section summarizes some possible methodologies to derive the performance parameters of IP packet transmission over the MSS links. In this section, the MSS link or satellite section refers to the satellite link between the satellite access terminals at both mobile and gateway earth stations via a satellite as depicted in Fig. 1 and Fig. 2 of Recommendation ITU-R M.1636.

2 IP packet error ratio

The IP packet error ratio (IPER) is defined, in ITU-T Recommendation Y.1540, as the ratio of total errored packets to the total successful and errored packets. The errored packet is defined such that the binary contents of the delivered packet information field do not conform exactly to those of the originated packet or that one or more of the header field(s) of the delivered packet(s) is (are) corrupted.

It would be possible to theoretically derive IPER caused by errored packets over the MSS link portion considering digital transmission performance parameters that are stipulated for MSS links. If the random error occurrence can be assumed, IPER, that is the probability that at least one bit will become errored for a packet, can be statistically derived as:

$$IPER = 1 - (1 - BER)^{Packet\_size \times 8} \qquad (1)$$

where:

BER: bit error rate after applying the possible forward error correction (FEC) scheme

Packet_size: size of IP packet in bytes transferred by the MSS link.

For example, Recommendation ITU-R M.1476 stipulates that a performance objective for the MSS forming part of the ISDN defines that the BER should be less than 9×10^{-7}. When an IP packet size of 1 500 bytes is assumed, the IPER for the system conforming to the ITU-R Recommendation becomes up to 1×10^{-2}.

It is generally considered that there would be a trade-off relation between the reducing packet error and reducing transfer delay. Such a trade-off depends on the system design.

For error-sensitive applications, ITU-T Recommendation Y.1541 stipulates that the performance objective for IPER is provisionally less than 1×10^{-4}. To comply with the ITU-T Recommendation for the error-sensitive QoS classes, one possible approach is to apply an ARQ scheme to the MSS link to improve IPER at the expense of an increase in IP packet transfer delay.

For delay-sensitive applications, no retransmission technique such as ARQ can be applied. Therefore, IPER is determined only by performance of the satellite channel, that is, *BER*, and the size of each IP packet, *Packet_size*, that should carefully be determined through the system design.

3 IP packet loss ratio

The IP packet loss ratio (IPLR) is defined, in ITU-T Recommendation Y.1540, as the ratio of total lost packets to the total transmitted packets. The causes for the loss of IP packets are, for example, misdirection of the IP packet due to the inconsistent update of routing tables, overflow of buffers at routers or packet transmission equipment, overload of routers, and so forth. Considering the MSS link section of the IP packet communication path, the primal cause, that is, the bottleneck for the IP packet loss would be the overflow of packets from the buffer at the interface of the satellite transmission equipment to connect the MSS link to the terrestrial section.

The ratio of IP packet loss due to the overflow of packets from the transmission buffer is dependent on the following parameters:

– IP packet arrival process;
– size of transmission buffer at an interface to connect the MSS link to the terrestrial section;
– scheme for IP packet scheduling over the MSS link.

It is widely recognized that the arrival process of the IP packet data application is bursty in nature. For example, a traffic model is defined for web browsing for the purpose of evaluating radio transmission technologies in mobile telecommunications applications[1]. The model employs the so-called on-off packet arrival model. The on- and the off-periods correspond to a series of packets for a file download and the reading time of the file, respectively. The model is a good representation of the web browsing activities; however no analytical approach has been reported nor established so far to evaluate the performance of IP packet data applications under the complicated packet data traffic model with the bursty nature. One possible approach to evaluate the *IPLR* under the bursty traffic is to apply the queuing simulation with appropriate assumption of the above parameters.

[1] UMTS TR 101 112, "Selection procedures for the choice of radio transmission technologies of the UMTS," April 1998.

Assuming the Poisson process for IP packet arrival and finite buffer with a FIFO scheduler, another possible approximation for the ratio of packet loss, *IPLR* due to buffer overflow can theoretically be derived by the analysis of the *M*/*M*/*1*/*K* queue as follows:

$$IPLR = \rho^K \Big/ \sum_{n=0}^{K} \rho^n \tag{2}$$

where:

- K: buffer size in terms of the number of packets
- ρ: traffic intensity, that is $\rho = \lambda \cdot h$
- λ: packet arrival rate (number of packets per second)
- h: average holding time for packet transmission.

4 IP packet transfer delay (IPTD)

IP packet transfer delay (IPTD) is the total transmission delay for an end-to-end IP connection. For IPTD, it is necessary for the end-to-end connection to be allocated properly to all sections that form the end-to-end connection. For the MSS link section, $IPTD_{sat}$ is defined in Recommendation ITU-R M.1636 as follows:

$$IPTD_{sat} = \sum_{n=1}^{N+1} \left\{ T_{n,propagation} + T_{n,processing} \right\} + T_{buffer} \tag{3}$$

where:

- N: number of retransmissions by ARQ
- $T_{n,propagation}$: propagation delay of an MSS link for the n-th time transmission
- $T_{n,processing}$: processing delay for packet transmission on the satellite channel and adaptation for the n-th time transmission on an MSS link
- T_{buffer}: buffering delay at an interface to connect the MSS link to a terrestrial section.

Typical values for the IP packet propagation delay, $T_{n,propagation}$ are provided in Recommendation ITU-R M.1636 for various MSS systems together with the delay variation. For non-GSO (MEO) MSS systems, $T_{n,propagation}$ is 69 ms at the sub-satellite point and is 103 ms at the edge of satellite coverage. Processing delay, $T_{n,processing}$ is dependent on the MSS system characteristics for IP packet transmission such as:

– scheme of forward error correction;

– frame structure for the MSS link;

– packet length and packet transmission rate (bit/s);

– scheme for scheduling of IP packet on the MSS link.

These parameters can be obtained from the performance objectives of the MSS physical and link layers. Buffering delay, T_{buffer} can be derived by queuing analysis or some type of simulation study if an appropriate model can be assumed for the packet arrival process, size of the buffer and the scheme for scheduling and framing of the IP packet on the MSS link.

In ITU-T Recommendation Y.1541, the performance objective for the packet transfer delay is defined as the mean IPTD.

For delay-sensitive applications, where no packet retransmission is allowed, the mean value of $IPTD_{sat}$, $\overline{IPTD_{sat}}$ can simply be derived by summing $T_{n,propagation}$, $T_{n,processing}$ and T_{buffer}.

For error-sensitive applications, $\overline{IPTD_{sat}}$ depends on the employed ARQ scheme.

For stop-and-wait ARQ, $\overline{IPTD_{sat}}$ can be derived as follows:

$$\overline{IPTD_{sat}} = \bar{x} + \frac{\lambda \overline{x^2}}{2(1-\lambda\bar{x})} - (T_{propagation} + T_{ack,processing}) \tag{4}$$

$$\bar{x} = \frac{T_D}{1-p} \tag{5}$$

$$\overline{x^2} = \frac{T_D^2(1+p)}{(1-p)^2} \tag{6}$$

where:

- λ: packet arrival rate (packets/s)
- p: IP packet error ratio, that is $IPER$
- T_D: round trip delay for the MSS link,

  $T_D = 2T_{propagation} + T_{data,processing} + T_{ack,processing}$

  - $T_{propagation}$: average propagation delay over the MSS link
  - $T_{data,processing}$ $T_{ack,processing}$: average processing delay for a data packet and an acknowledgement packet, respectively.

For go-back-N ARQ, $\overline{IPTD_{sat}}$ can be derived as follows:

$$\overline{IPTD_{sat}} = \bar{x} + \frac{\lambda \overline{x^2}}{2(1-\lambda\bar{x})} + (T_{propagation}) \tag{7}$$

$$\bar{x} = T_{data,processing} + T_D \cdot \frac{p}{1-p} \tag{8}$$

$$\overline{x^2} = T_{data,processing}{}^2 + 2T_{data,processing} \cdot T_D \cdot \frac{p}{1-p} + T_D^2 \cdot \frac{p(1+p)}{(1-p)^2} \tag{9}$$

Lastly, for selective-repeat ARQ, $\overline{IPTD_{sat}}$ can be derived as follows:

$$\overline{IPTD_{sat}} = \bar{x} + (T_D - T_{data,processing}) \cdot \frac{p}{1-p} + \frac{\lambda \overline{x^2}}{2(1-\lambda\bar{x})} + T_{propagation} \tag{10}$$

$$\bar{x} = \frac{T_{data,processing}}{1-p} \tag{11}$$

$$\overline{x^2} = T_{data,processing}^2 \frac{1+p}{(1-p)^2} \tag{12}$$

5 IP packet delay variation

The derivation of IP packet delay variation (IPDV) is shown in Appendix IV of ITU-T Recommendation Y.1541, where various contributors should be taken into account. In addition, the fluctuation of the propagation delay also affects IPDV in IP packet data transmission over non-GSO MSS links, where the typical value for the variation of the propagation delay can be found in Recommendation ITU-R M.1636. Overall IPDV over the MSS link should take the aforementioned factors into account, and it is too complicated to be derived by queuing analysis. One possible approach for the derivation of IPDV over the MSS link is to generate a distribution function for IPTD by the measurement of existing typical MSS systems or by simulation.

Annex 3

Guidelines for optimization of performance for IP packet data applications in the MSS

1 Introduction

In today's Internet communications, TCP is the most dominant transport layer protocol over the IP layer. The performance of the IP packet transmission using TCP is primarily determined by the transmission bandwidth and transmission delay. When the propagation delay becomes large, such as for satellite communications links, the throughput performance of TCP would be significantly degraded. For example, if a round trip time of 200 ms and a window size of 8 kbytes are assumed, the throughput of TCP is limited to 310 kbit/s no matter how large the bandwidth of the access network.

To handle the large transmission delay in the MSS links, the window scale option, which enables TCP to use a window size larger than 64 kbytes, is widely used among the various TCP kernel software modules in the recent operation systems. This section describes the TCP window scale tuning for large propagation delay.

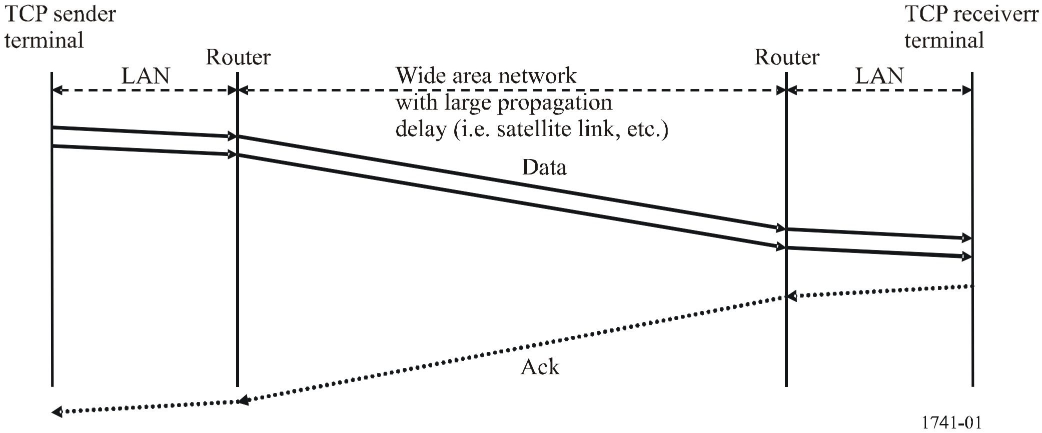
2 Problems of end-to-end communication with large propagation delay

TCP performs flow control based on the sliding window mechanism between end terminals on an end-to-end basis as shown in Fig. 1. Flow control of data is performed by TCP kernel software on the PCs or workstations. The window size is used to control the data flow and it specifies the data amount which can be sent without being acknowledged. The maximum size of the TCP window used is generally 64 kbytes and 8 kbytes.

This small window size has serious disadvantages if the propagation delay of the link, e.g. 80 ms delay between the east coast of the United States and Japan via an optical fibre cable is large. As illustrated in Fig. 1, a sending TCP terminal cannot send data even if the access link is idle because ACK is not sent back for many hours.

FIGURE 1

Sequence of end-to-end flow control with insufficient window size

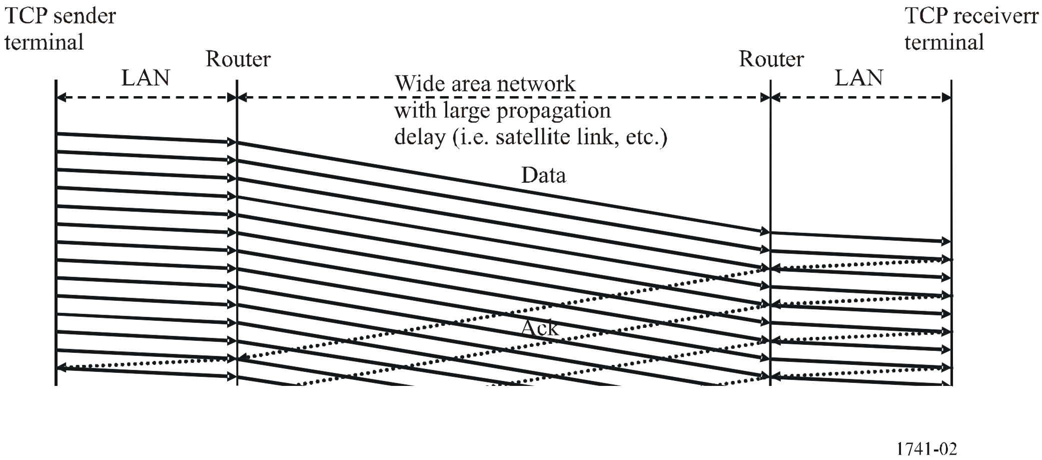


3 Approach to enhance the TCP throughput performance

TCP parameter tuning is one of the solutions to enhance the TCP throughput performance. The key TCP parameter is the maximum TCP window size. Especially in the case where the satellite link is part of the wide area network, a large window size over 64 kbytes is required. However, the traditional TCP software does not permit a maximum window size over 64 kbytes. To handle this problem, the TCP extensions are currently specified for higher performance under the larger delay environment. The window scale option is one of the TCP extensions for this purpose. It enables high TCP throughput performance even when using satellite links. As illustrated in Fig. 2, a sending TCP terminal can always send the data under the large propagation delay environment.

FIGURE 2

Sequence of end-to-end flow control with sufficient window size



4 Estimated TCP throughput

Assuming there is no TCP segment loss in the wide area network, the estimated TCP throughput is calculated by the following equation:

"Estimated TCP throughput" =

"Maximum TCP window size"/("RTT (round trip time) between
TCP sender and receiver" + "TCP segment send time at the sender").

Table 2 shows the relation between RTT and the estimated throughput under the two types of TCP maximum window sizes. The values are calculated under the condition that the MTU (maximum segment size) is 1 460 bytes (i.e. maximum transmission unit is 1 500 bytes) and the sender access line speed is 400 kbit/s as the bottleneck bandwidth (i.e. TCP segment send time at sender is 0.03 s).

TABLE 2

Estimated TCP throughput

	Maximum window size = 64 kbytes	Maximum window size = 512 kbytes
RTT = 50 ms	6.55 Mbit/s	52.4 Mbit/s
RTT = 360 ms	1.34 Mbit/s	10.8 Mbit/s
RTT = 1 000 ms	0.51 Mbit/s	4.07 Mbit/s
RTT = 2 000 ms	0.26 Mbit/s	2.07 Mbit/s

From the result of Table 1 in Annex 1, the TCP maximum window size needs to be tuned according to the RTT between the TCP sender and receiver. It should be noted that the estimated TCP throughput must align with the bottleneck bandwidth (i.e. bandwidth of the MSS link) if the bottleneck bandwidth is lower than the calculated results. Therefore, if the TCP throughput in Table 2 is lower than the bottleneck bandwidth, that is the bandwidth of the satellite access link in MSS, TCP maximum widow size should be larger so that the bandwidth can be efficiently utilized by the IP packet transmission.

5 Another approach to enhance TCP/IP performance

Another possible approach for the TCP/IP performance enhancement in the satellite transmission channels is to use the segment splitting method. The method splits the end-to-end TCP sessions into two or three segments so that the satellite link segment with large transmission delay will not deteriorate the end-to-end TCP/IP performance. ITU-R has studied this segment splitting approach and the results of this study are incorporated in Recommendation ITU-R S.1711. Descriptions in Recommendation ITU-R S.1711 with the experimental test results provide a detailed understanding of the approach and its effects on TCP/IP performance enhancement.