International Telecommunication Union

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

G.114

Amendment 2

(11/2009)

SERIES G: TRANSMISSION SYSTEMS AND MEDIA, DIGITAL SYSTEMS AND NETWORKS

International telephone connections and circuits – General Recommendations on the transmission quality for an entire international telephone connection

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One-way transmission time

Amendment 2: New Appendix III – Delay variation on unshared access lines

Recommendation ITU-T G.114 (2003) – Amendment 2

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Recommendation ITU-T G.114

One-way transmission time

Amendment 2

New Appendix III – Delay variation on unshared access lines

Summary

Appendix III of Recommendation ITU-T G.114 provides an introduction on the effect of different IP services on lines with limited bandwidth (e.g., DSL). It explains the mechanism of serialization delay, gives a (very general) overview over prioritization and shows how the maximum delay variation due to concurrent traffic can be calculated. The calculations shown in this appendix are valid for unshared lines only, shared lines are excluded.

The intention of this appendix is to make standards developers aware of this behaviour.

Source

Amendment 2 to Recommendation ITU-T G.114 (2003) was agreed on 12 November 2009 by ITU-T Study Group 12 (2009-2012).

FOREWORD

The International Telecommunication Union (ITU) is the United Nations specialized agency in the field of telecommunications, information and communication technologies (ICTs). 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.

In some areas of information technology which fall within ITU-T's purview, the necessary standards are prepared on a collaborative basis with ISO and IEC.

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New Appendix III – Delay variation on unshared access lines

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

III.1 Introduction

For some time now, the coverage of broadband access for customers has been growing higher and higher. This broadband access is used for different services, starting with Internet, and nowadays, increasingly TV and voice services. Packet-based networks offer high flexibility to deliver all of these services over the same network. If more than one service is used at the same time or if one service uses more than one session at a time, there is the possibility of interference. One very real effect will be the influence of other services/sessions on VoIP-media traffic.

III.2 Serialization delay

Serialization delay of a packet is the time it takes to clock every bit of a packet onto the line. A packet ready to be sent will normally be put in a playout buffer, from where it will be clocked onto the line at the physical line speed (see Figure III.1).

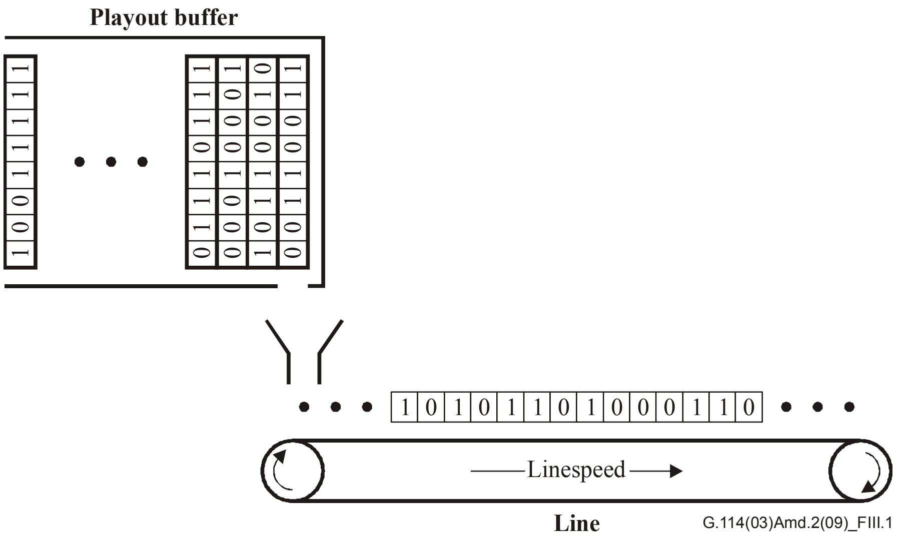

Figure III.1 – Serialization delay

The equation to calculate the serialization delay $t_{serialisation}$ is as follows:

$$t_{serialisation}[s] = \frac{Packetsize[\text{bit}]}{Linespeed[\text{bit/s}]}$$

With

Packetsize = size of a packet on the physical layer

Linespeed = line speed on the physical layer

Table III.1 shows some example calculations.

NOTE – It is recommended to do this calculation on the physical layer even if it could be done on any other layer, as long as the packet size and the line speed are calculated for the same layer. It has to be taken into account that the packet size needs to represent the size of a packet, including all headers and trailers (for further calculations it may also be necessary to include the minimal distance between two packets).

Table III.1 – Example of serialization delays with different line speeds and packet sizes

Line speed	Packet size	Serialization delay
1 Gbit/s	1500 bytes	0.012 ms
1 Gbit/s	200 bytes	0.0016 ms
100 Mbit/s	1500 bytes	0.12 ms
100 Mbit/s	200 bytes	0.016 ms
10 Mbit/s	1500 bytes	1.2 ms
10 Mbit/s	200 bytes	0.16 ms
1 Mbit/s	1500 bytes	12 ms
1 Mbit/s	200 bytes	1.6 ms
100 kbit/s	1500 bytes	120 ms
100 kbit/s	200 bytes	16 ms

III.3 Prioritization

The general concept of prioritization is that traffic with higher priority is favoured against traffic with lower priority on the same line. Prioritization is important in the case where a bandwidth limitation exists (more input capacity than output capacity) or in case of congestion (these two effects can be related). Prioritization is a strong method for queue management (strict priority), which means that this traffic is prioritized in any case; or a weaker bandwidth allocation (like a weighted fair queuing) in order to allow lower "prioritized" traffic to pass even in the case of congestion.

Normally, VoIP media traffic is in the highest priority class which uses strict priority queues (in fact, internal network control traffic is prioritized even higher). For the remainder of the traffic, there are several priority classes possible, which normally use fair queuing.

Figure III.2 shows a functional diagram of a typical prioritization algorithm in network equipment.

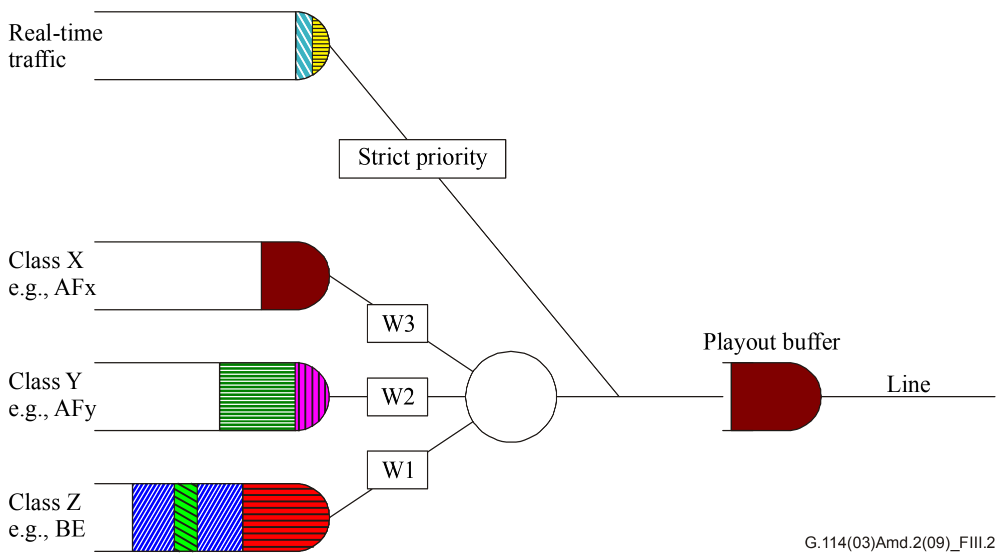

Figure III.2 – Prioritization principle

Classes X, Y and Z are three differently prioritized classes (in Figure III.2, classes X and Y are two different assured forwarding (AF) classes, while class Z is a best effort (BE) class). In this case, the prioritization is according to different weights (W1, W2 and W3) of the classes, but every class will get its time slot to send (based on the weighting). This means that it is possible that a lowest-prioritized packet from the queue of class Z can be sent to the playout buffer, even if there are packets in the higher prioritized queues of class X and Y.

Since the real-time class is allocated strict priority, no packet out of queues X, Y and Z can be sent to the playout buffer if there is a packet in the real-time queue (this implicitly means that there has to be some sort of bandwidth control for the real-time class to avoid a blockage of all other traffic).

In the playout buffer, there is no prioritization. Packets in the playout buffer will be sent to the line in the order they arrived (FIFO, first in, first out).

If the playout buffer is full, not even a packet from the real-time queue can be sent to it. The prioritized packet from the real-time queue has to wait until the packet in the playout buffer is sent to the line.

With the equation for the serialization delay, the time taken can be calculated. This effect leads to delay variation for real-time traffic.

In a usual priority implementation, the playout buffer has the capability to hold two packets; this means that, in the worst case, two low-prioritized packets have to be sent before a high priority packet can be sent.

The equation for the maximum delay variation, $t_{delayvariation}$, due to this effect will be the one for serialization delay multiplied by the number of packets which the playout buffer can hold.

NOTE 1 – In difference to the equation for serialization delay, where the actual packet size of the packet of interest has to be taken, in the following equation for the maximum delay variation, the maximum packet size possible on the link has to be taken.

$$t_{delayvariation}[\mathrm{s}] = \frac{NbrofPackets * MaxPacketsize[\mathrm{bit}]}{Linespeed[\mathrm{bit/s}]}$$

With

NbrofPackets	=	maximum number of packets in the playout buffer
MaxPacketsize	=	maximum size of a packet on the link (physical layer)
Linespeed	=	line speed on the physical layer

NOTE 2 – A typical maximum packet size for IP networks is around 1500 bytes at the IP-layer.

III.4 Measurement examples

Figures III.3 to III.5 show real IP packet delay variation measurements for a VoIP call between two DSL access examples, customer A with 6400/640 kbit/s access speed, and customer B with 4608/576 kbit/s access speed.

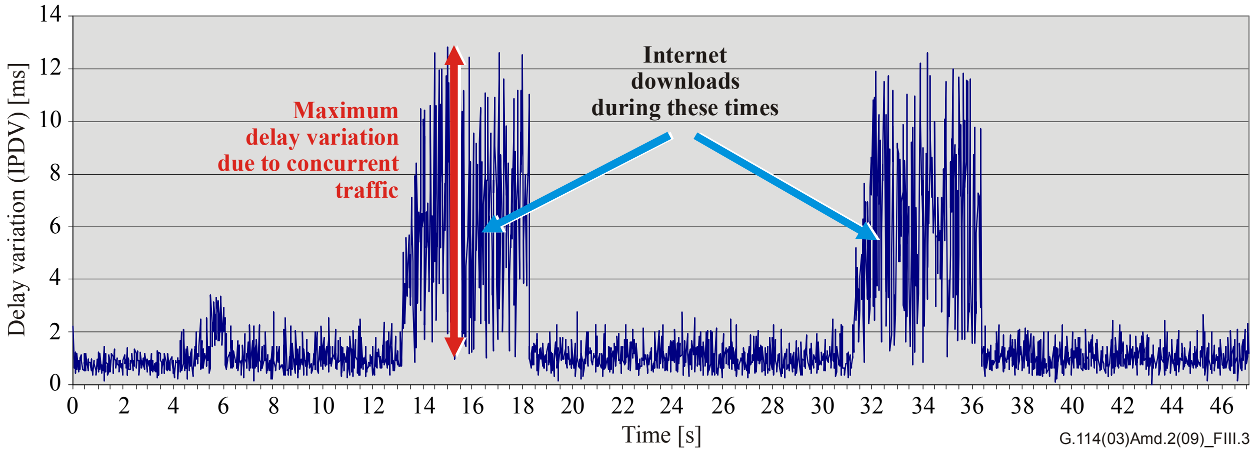

Figure III.3 – Delay variation (IPDV) of a call A to B, with parallel download at B

Figure III.3 shows the delay variation (IPDV) over time measured at the customer B side, with two intermittent Internet downloads also on the B side. According to the equation for the serialization delay, one Internet packet (1500 bytes at IP layer → approximately 1696 bytes on the physical layer in this case) will have a serialization delay of 2.94 ms (downstream bit rate B: 4608 kbit/s). Since the maximal delay variation measured is much higher (nearly 12 ms, taken as the difference of the maximum IPDV and the IPDV without Internet download), it can be assumed that the playout buffer of the network equipment involved in the prioritization towards the DSL line holds up to four IP packets. With a better implementation of the prioritization algorithm, IPDV could be reduced by 9 ms.

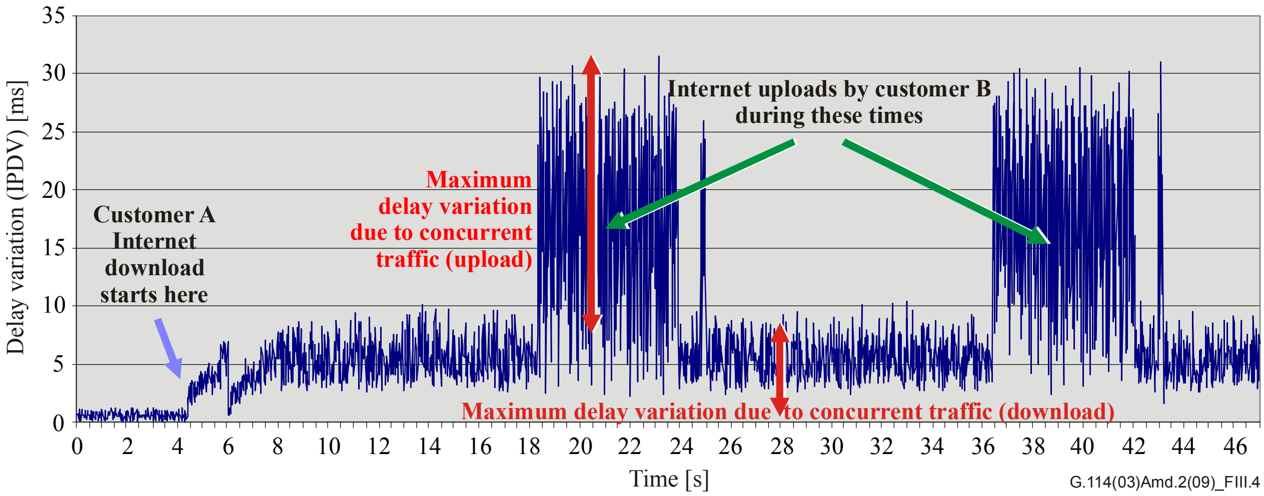

Figure III.4 – Delay variation (IPDV) of a call B to A with parallel download at A and upload at B

Figure III.4 shows the delay variation (IPDV according to [i1]) over time measured at the customer A side, with a continuous download at the A side (starting at 4s) and two intermittent uploads at the B side. The serialization delay of one Internet packet (1500 bytes at IP layer → approximately 1696 bytes on the physical layer in this case) on the A side will be 2.12 ms (downstream bit rate A: 6400 kbit/s). Since the maximal delay variation measured is much higher (about 8 ms, taken as the difference of the maximum IPDV without upload and the IPDV without up- and download), the conclusion from the previous measurement is confirmed: the playout buffer of the network equipment involved in the prioritization towards the DSL line holds up to four IP packets.

The serialization delay of one Internet packet (1500 bytes at IP layer → approximately 1696 bytes on the physical layer in this case) on the B side will be 23.5 ms (upstream bit rate B: 576 kbit/s). This is the delay variation measured during the two intermittent periods of upstream traffic (taken as the difference of maximum IPDV during up- and download and maximum IPDV during download). This means that the playout buffer of the customer router holds only one IP packet, which is the optimum.

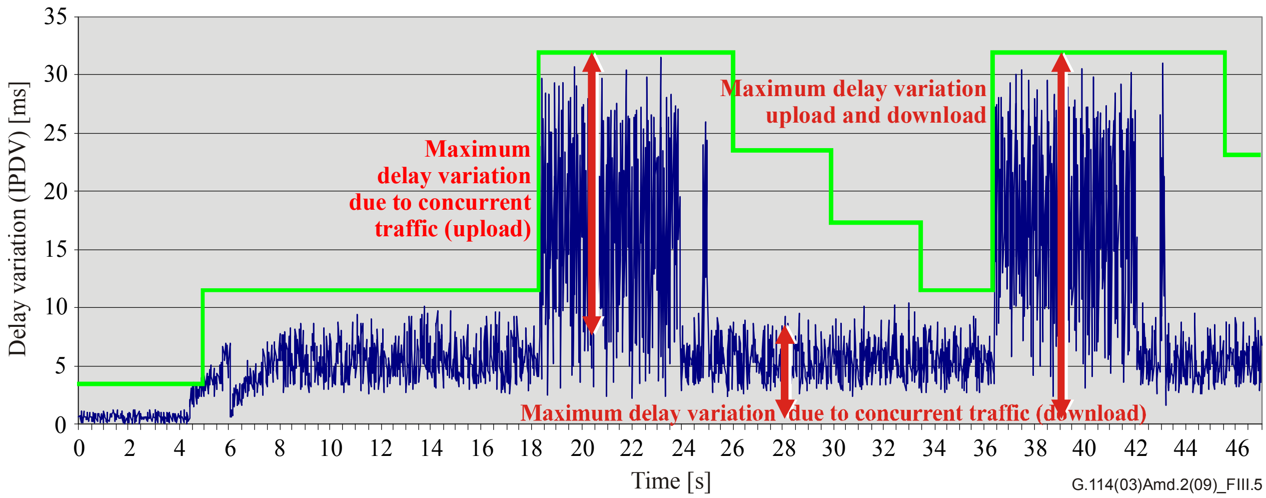

Figure III.5 – End-to-end (audio) delay variation

Figure III.5 shows a possible end-to-end (audio) delay variation. This delay variation depends on the dejitter buffer behaviour, so the stepped line is an example. Figure III.5 also shows that if there is delay variation on both sides of a connection, the resulting delay variation will be the sum of the two delay variations.

III.5 Theoretical considerations

Unfortunately, there are not many options to overcome the problem of exceeding delay variation on access lines.

- Using ATM with different PVCs per prioritization class (and a playout buffer per PVC): With this solution, the maximal packet size on the link will be 53 bytes (ATM frame size) instead of over 1500 bytes. Unfortunately, ATM will no longer be an option as it will be progressively phased.
- Fragmentation of lower prioritized packets: If the maximum packet size for lower prioritized traffic is set to one half (e.g., from 1500 bytes to 750 bytes), the maximum delay variation will also be only half of the delay variation without fragmentation. This approach has two major disadvantages: there will be more overhead, leading to less net bandwidth, and there will be additional load on the network equipment which has to do the fragmentation/defragmentation.
- If the physical bandwidth is enhanced, the maximum delay variation will decrease. Obviously, the enhancement of the physical bandwidth is often not an appropriate solution due to economic constraints.

- Tell the customer not to use the Internet or any other service while using VoIP. This would not help much, because there is still parallel signalling traffic used for the VoIP connection also causing delay variation, and also no operator will tell a customer that he cannot use all IP services in parallel.

III.6 Conclusions

The mechanism shown in this appendix leads to the conclusion that, for VoIP services, the access parts of a connection lead to substantial delay variation, which has to be taken into account for network planning purposes.

Furthermore, if there is concurrent traffic on both sides of a VoIP connection (upstream on one side, downstream on the other side), the resulting delay variation is the sum of both delay variations and, consequently, a normal jitter buffer on the receiver side will produce an additional delay of at least the sum of both delay variations.

The possible theoretical solutions are often not applicable in practice.

For an operator, it is necessary to find a balance between coverage (which is given with the minimum access bandwidth needed for VoIP) and voice quality (which is the maximal delay variation allowed on a connection).

If delay variation limits in planning guidelines are set too low, many customers will never be able to access VoIP services.

Therefore, it is important to find a solution for planning guidelines which allows lower bandwidth accesses (→ higher delay variation limits) without relaxing the delay variation limits for high bandwidth access.

III.7 Abbreviations and acronyms

This appendix uses the following abbreviations and acronyms:

ATM Asynchronous Transfer Mode

DSL Digital Subscriber Line

FIFO First In, First Out

IPDV IP Packet Delay Variation

PVC Permanent Virtual Connection

VoIP Voice over Internet Protocol

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