International Telecommunication Union

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TELECOMMUNICATION STANDARDIZATION SECTOR OF ITU

Y.1540

Amendment 1

(03/2009)

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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Internet protocol data communication service – IP packet transfer and availability performance parameters

Amendment 1: New Appendix VII – Packet performance parameters for optimization of stream repair techniques and new Appendix VIII – IP-layer capacity framework

Recommendation ITU-T Y.1540 (2007) – Amendment 1

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Recommendation ITU-T Y.1540

Internet protocol data communication service – IP packet transfer and availability performance parameters

Amendment 1

New Appendix VII – Packet performance parameters for optimization of stream repair techniques and new Appendix VIII – IP-layer capacity framework

Summary

Amendment 1 to Recommendation ITU-T Y.1540 introduces Appendices VII and VIII.

Appendix VII builds on the fundamental Y.1540 definitions and concepts to describe a new set of performance parameters. The objective of the new parameters is to provide information relevant to the design and configuration of higher-layer (application-layer) techniques to compensate for packet loss due to various causes (including errors and delay variation). Thus, the design and/or optimization of application-stream repair techniques should be simplified if these new metrics for packet performance assessment meet their goal.

Appendix VIII builds on the definitions in Recommendation ITU-T Y.1540 and provides a complementary framework of performance parameters. This framework defines parameters related to IP characteristics of an exchange link and its extension into network paths. One of the parameters in this framework is the IP-layer available capacity which describes the amount of network capacity that is available for network applications to utilize without causing congestion.

Source

Amendment 1 to Recommendation ITU-T Y.1540 (2007) was agreed on 19 March 2009 by ITU-T Study Group 12 (2009-2012).

FOREWORD

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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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Recommendation ITU-T Y.1540

Internet protocol data communication service – IP packet transfer and availability performance parameters

Amendment 1

New Appendix VII – Packet performance parameters for optimization of stream repair techniques and new Appendix VIII – IP-layer capacity framework

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

VII.1 Introduction

IP-layer performance parameters have many uses, with network monitoring and trouble identification being one class of use. The parameters are also used as the basis of service level agreements (SLA). Both the aforementioned uses describe packet transfer as a characterization of the network which provided the UNI-UNI transport.

There is a second perspective: IP-layer performance parameters also characterize networks in terms which can be relevant to the application designer. Although many of the parameters used in network monitoring are useful to application designers, there are likely to be unique parameters for each use case. Figure VII.1 illustrates the two different perspectives, or use cases for IP performance parameters.

Recommendation ITU-T Y.1540 defines performance and availability parameters for IP-based networks. It defines primary and secondary packet transfer outcomes and a range of packet performance parameters based on these outcomes, including the IP service availability function.

This appendix builds on the fundamental Y.1540 definitions and concepts to describe a new set of performance parameters. The objective of the new parameters is to provide information relevant to the design and configuration of higher-layer (application-layer) techniques to compensate for packet loss due to various causes (including errors and delay variation). Thus, the design and/or optimization of application-stream repair techniques should be simplified if these new metrics for packet performance assessment meet their goal.

This appendix begins with a short background on application-layer stream repair techniques. It then goes on to offer a very simple model intended to be applicable to many different repair techniques and then defines the performance parameters needed to measure network performance and employ the model.

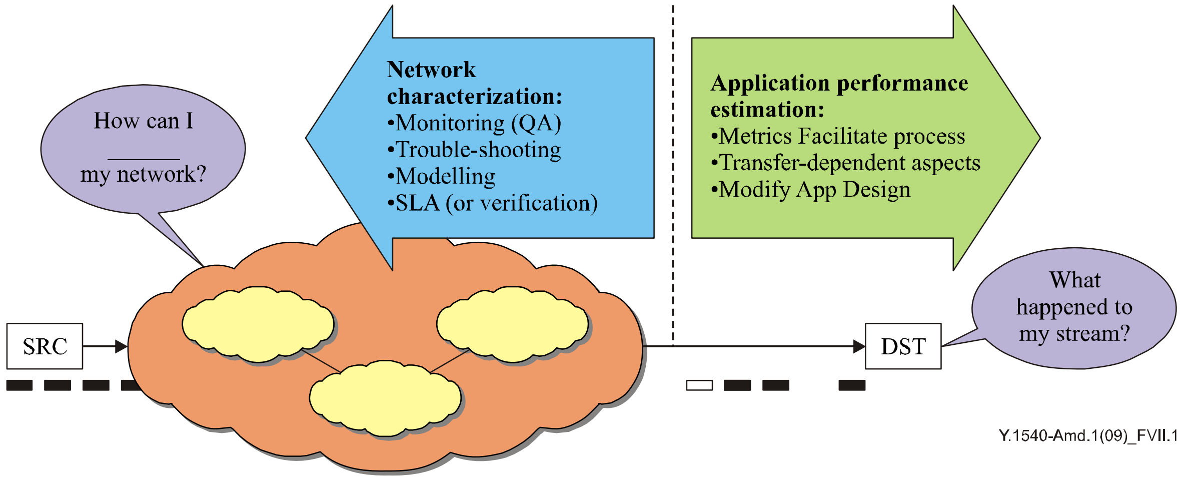

Figure VII.1 – Two different use cases for IP performance parameters

The usual procedure is to introduce new metrics as informative appendices, so that potential users have the opportunity to evaluate them prior to their incorporation as normative parameters in the body of the Recommendation. These new metrics are following the informative-first path to incorporation in Recommendation ITU-T Y.1540 and other Recommendations, as needed.

VII.2 Short description of application-layer stream repair techniques

There are three main types of application-layer techniques to compensate for packet transport impairments. We focus on continuous real-time or near-real-time applications (audio, video) that are non-elastic – information delivery must take place according to a predetermined time schedule, and not the class of elastic data transfer applications usually served by TCP and its reliable octet stream transfer services.

Forward error correction (FEC): This is a technique where streams of packets are organized into blocks prior to transfer. There are calculations performed on each block, and overhead packets added to the stream which the receiver can use to reproduce some fraction of the packets in the block if they are lost, or successful but delayed or corrupted in transport. Typical overhead represents 5% to 20% of the information block. In an *ideal* FEC scheme, the number of lost packets that can be corrected is *equal* to the number of overhead packets. The key aspects of this scheme are:

- The size of the information **block**, in packets and time;
- The amount of **overhead** packets relative to the information block, expressed as a fraction.

Automatic repeat-reQuest (ARQ): In this technique, there is a reverse communication channel available where the receiver, having *detected* that specific individual packets are lost, delayed, or corrupted, can request retransmission (this is referred to as a selective ARQ). The lost packets are re-sent in time for them to take their place as the information is passed to higher layers for decoding and play-out. Transmission control protocol (TCP) has sometimes been modified to serve non-elastic streams in the role of ARQ. There is a waiting time for determining whether packets are simply delayed or lost, and this is similar to the information block used in FEC schemes. There may also be a limit on retransmitted packets which can accompany the primary stream in any time interval, and this is parallel to the overhead of FEC schemes. The ARQ technique can retransmit a number of lost packets in a block, equal to its limit on retransmission overhead. Note that the retransmitted packets will represent overhead on a subsequent block of information packets, but the concept still applies.

Thus, the ARQ and FEC techniques can both be described using the same basic variables of information block size and overhead size.

Application-layer error concealment: This is a technique where decoders attempt to compensate for lost or corrupted information, using a variety of application-specific techniques, some of which have been standardized. The applicability of the simple model (derived below) to this class of techniques is for further study.

VII.3 Simple model of application-layer stream repair techniques

Each stream of application-layer packets is modelled as containing two categories of packets:

1) blocks of information packets;

2) overhead packets, associated with the information block.

The challenge to the repair technique designer is to choose the information block size in combination with the (maximum) amount of overhead packets that will be sufficient to compensate for a high percentage of packet network impairments (loss, excessive delay, and corruption), while working within the overall packet transfer capacity limits of the system and delivering sufficient quality in the application stream.

The new performance parameters should aid these decisions.

VII.4 New performance parameters to characterize stream repair variables

The following definitions for evaluation of a set of consecutive packets are candidates for future inclusion in the body of Recommendation ITU-T Y.1540. First, we define a new outcome, and then a new parameter based on that outcome.

Outcome

IP packet impaired interval outcome: An IP packet impaired interval outcome occurs for a set of packets observed during time interval T_I at ingress MP_0 when the ratio of impaired packet outcomes at egress MP_i to total packet outcomes in the interval exceeds s2. Impaired packet outcomes are the sum of the following outcomes:

- Lost packet outcomes, using a T_{max} associated with T_I and the nominal transfer time, and possibly equal to the minimum packet transfer delay for the population of interest plus T_I. This would include packets that are subject to excessive queuing as well as those that never arrive.
- Errored packet outcomes.

There are no provisional values set for the time interval T_I and the threshold s2. Instead, the analysis may involve a range of values for interval T_I and threshold s2. The length of the IP packet payload should also be specified, as this influences the serialization time and therefore the time interval occupied by a block of packets.

Parameter

Ideally, we would like to know the probability that a given packet interval (information block, b, plus overhead, x) will contain more than x impairments.

$$P(b+x, x) = p, \text{ or } P(T_I, s2) = p$$

The measurement of the impaired packet outcomes occurring in a *population of interest* should provide an empirical assessment of the probability during available time.

IP packet impaired interval ratio (IPIIR): An IP packet impaired interval ratio is the ratio of the IP packet impaired interval outcomes to total intervals in a population of interest.

Figure VII.2 below gives an example that should accompany the parameter section, where T_I= 9 packets and s2=3 packets.

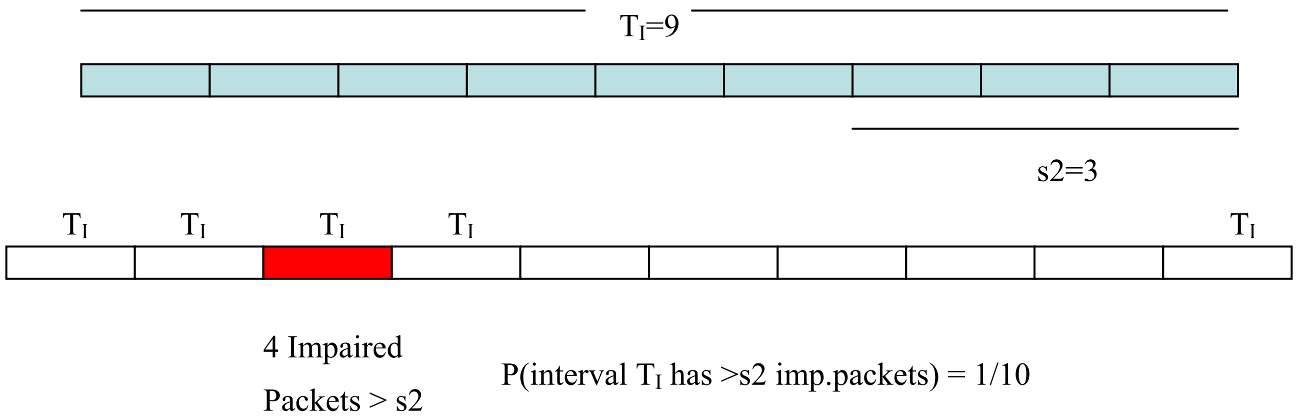

Figure VII.2 – Illustration of IPIIR performance parameter

VII.5 Discussion

The time intervals T_I MAY be overlapping, to allow assessment of different interval vs. impairment alignments (sliding interval analysis). There is an issue for fixed, non-overlapping intervals, that the actual information block + overhead may experience worse performance owing to the difference in alignment.

There are two approaches to characterizing packet streams to determine the optimum combination of stream repair variables:

1) using (multiple) arbitrarily-established packet intervals (in terms of time or number of packets), as done above;
2) counting intervals of consecutive impaired packets and intervals of unimpaired packet transfers.

The approach of counting consecutive intervals appears to have flexibility not available with evaluation based on fixed intervals; it can determine the actual size of impaired/un-impaired intervals in a stream and does not suffer from the interval alignment issue. However, summary parameters describing impaired/unimpaired interval lengths are independent from the actual sequence in which they occurred. This sequence of changes between impaired intervals and unimpaired intervals MAY be important. Also, the counter approach requires some way to evaluate whether the s2 threshold has been crossed, as this is essential to the definition of an impaired outcome. If more than one value of s2 is to be evaluated, then multiple passes through stored data may be needed.

In either case, the results can be expressed as probability or cumulative distributions over the dependent and independent variables, as the example below shows (Figure VII.3).

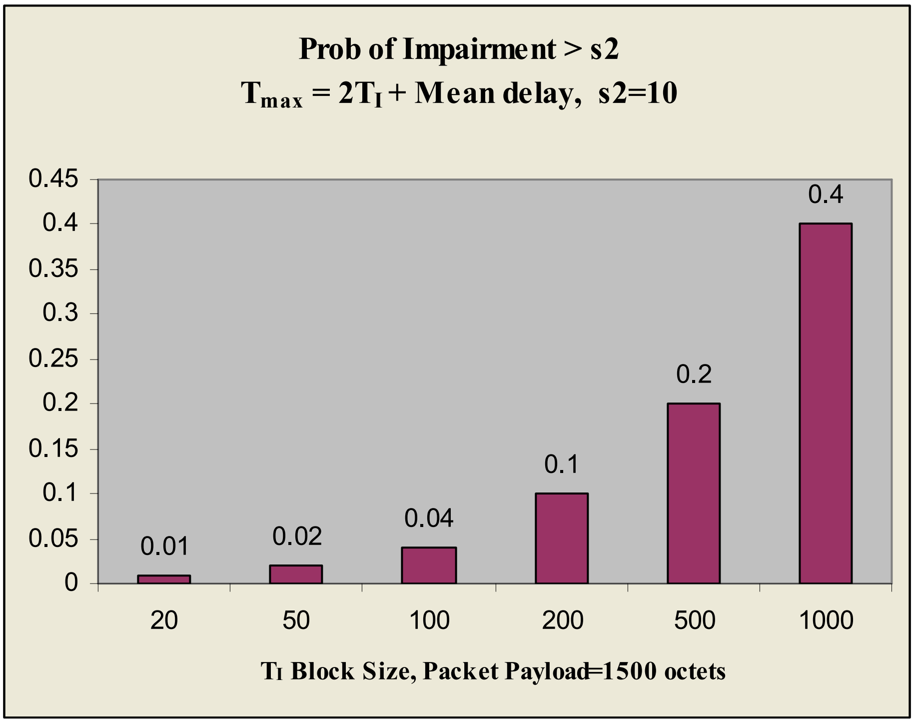

Figure VII.3 – Example plot of stream repair parameter results for a range of T_I block sizes, where s2 is fixed, packet size is fixed, and T_{max} varies based on T_I (expressed in units of serialization time) and the mean 1-way delay

VII.6 Additional considerations

Although network characterization using the parameters defined above may be useful, the application repair system details should be known to begin to predict the quality delivered to users. FEC and ARQ techniques produce different packet loss patterns when operating beyond their ability to perform complete loss correction. The typical block sizes associated with each technique are different, with ARQ often characterized by larger block sizes.

FEC schemes organize the information block and overhead packets in different ways (sometimes called one-dimensional or two-dimensional forms) with less sophisticated schemes having more sensitivity between the exact pattern of losses and their ability to correct the losses. The performance margin between simple FEC schemes and the ideal performing scheme predicted by the parameters above should be known to the designer and taken into account.

Some applications may use chains of the various techniques described above. For example, a system might use FEC or ARQ in combination with application-layer error concealment. In another example, there could be FEC used in one part of the path, with ARQ or a different FEC used in another part of the path, and finally employing application-layer error concealment.

Finally, the short-term performance parameters defined above may be useful in trouble-shooting by helping to identify the signatures of network problems, but this is for further study.

New Appendix VIII – IP-layer capacity framework

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

VIII.1 Introduction

IP-layer performance parameters such as packet transfer delay, inter-packet delay variation, packet error ratio, and packet throughput are vital when studying the performance of an IP network. Additional information is provided by performance indicators such as service availability or service-response time. These parameters have many uses such as for service-layer agreement verification, network monitoring, network anomaly detection and troubleshooting. Performance parameters on the IP layer can also be utilized to characterize the network in terms applicable to higher-layer applications, for example to adapt the bit rate of a media stream to reduce congestion.

Recommendation ITU-T Y.1540 describes the fundamental network building blocks as well as performance parameters for IP networks, such as the parameters mentioned above.

This appendix builds on the definitions in Recommendation ITU-T Y.1540 and provides a complementary framework of performance parameters. This framework defines parameters related to IP characteristics of an exchange link and its extension into network paths. One of the parameters in this framework is the IP-layer available capacity which describes the amount of network capacity that is available for network applications to utilize without causing congestion. Examples of the applicability of this framework are discussed in clause VIII.4.

VIII.2 IP-layer capacity framework

The IP-layer capacity framework defined in this appendix is designed to be in line with the framework provided by the IETF IPPM working group in [b-IETF RFC 5136].

The framework defines properties in terms of performance parameters of exchange links and paths (consisting of one or several exchange links in sequence) on the IP layer, as illustrated in Figures VIII.1 and VIII.2.

For a given population of interest, an exchange link is characterized by its IP-layer link capacity, the IP-layer used link capacity and the IP-layer available link capacity. See Figure VIII.1 for an example.

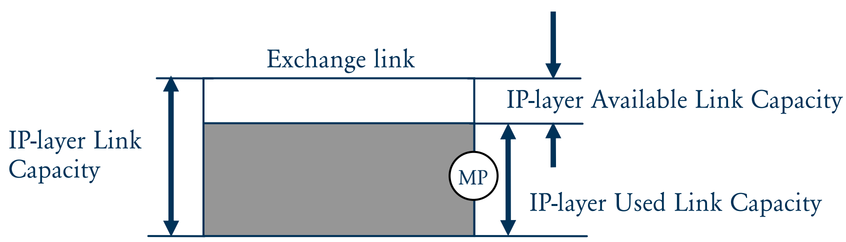

Figure VIII.1 – An exchange link and its corresponding properties defined in this framework

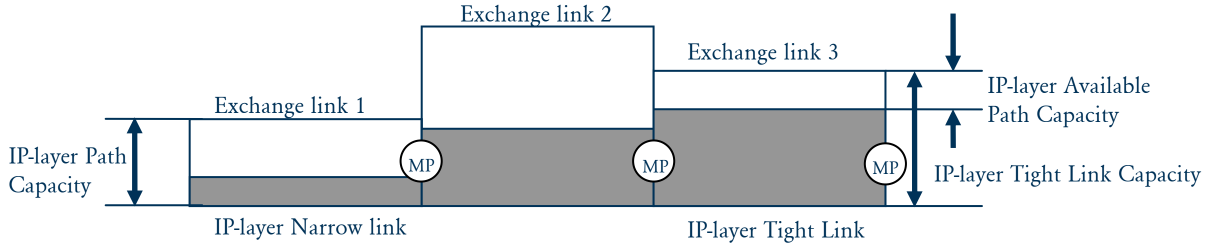

Figure VIII.2 – A network path and its corresponding properties defined in this framework

A path is a directed sequence of one or several consecutive exchange links. Two exchange links are connected via a host (e.g., a router). In the example depicted in Figure VIII.2, a path consisting of three exchange links is shown. The exchange links are numbered from 1 to 3, where the number is increasing in the direction from the source to the destination. Note that the direction of the path is significant. The properties of a path in one direction are typically not the same as in the opposite direction. This is a direct consequence of the fact the properties of an exchange link typically differ by direction. Notably, the IP-layer used link capacity, related to the network traffic, typically is not the same in the uplink as in the downlink.

The path-related properties in the framework are the IP-layer path capacity, the IP-layer tight link capacity and the IP-layer available path capacity.

Since the network traffic is dynamic, the parameters illustrated in the figures are time dependent. Further, the parameters represent a mean value over a specified time interval.

Measurement points, illustrated as circles, are located at hosts between two exchange links. As pointed out in the normative text of Recommendation ITU-T Y.1540, the exact location of the measurement point within a host is for further study.

Note that only IP packets that generate successful IP packet transfer outcomes contribute to the performance parameters. Further, note that a population of interest can refer to the IP packet size, the IP header size (e.g., depending on whether IPv4 or IPv6 is used) or a given diffserv class.

The framework consists of the following definitions.

VIII.2.1 IP-layer bits transferred

IP-layer bits are defined as eight (8) times the number of octets in all IP packets generating successful IP packet transfer outcomes at an egress measurement point, from the first octet of the IP header to the last octet of the IP packet payload, inclusive.

Note that this definition is similar to the definition of IP-layer bits in [b-IETF RFC 5136]. Also note that the definition of IP-layer bits is IP-version agnostic.

VIII.2.2 IP-layer link capacity

For a given population of interest, the IP-layer link capacity is:

$$C(t,\Delta t)=\frac{n_0(t,\Delta t)}{\Delta t}$$

where n_0 is the highest number of IP-layer bits that can be transmitted between two measurement points separated by an exchange link generating successful IP packet transfer outcomes at the egress measurement point during a specified time interval $[t, t+\Delta t]$.

VIII.2.3 IP-layer path capacity

The definition of IP-layer link capacity can be extended to an entire path. For a given population of interest, the IP-layer path capacity $C_P(t, \Delta t)$ during a specified time interval $[t, t + \Delta t]$ is defined as the smallest IP-layer link capacity along that path. That is, the IP-layer path capacity is

$$C_p(t,\Delta t) = \min_{i=1..n} C_i(t,\Delta t)$$

where C_i is the IP-layer link capacity of the exchange link number i (i=1..n) on the path between two measurement points.

Note that the exchange link with the smallest IP-layer link capacity is often called the narrow link in academic literature. The IP-layer link capacity of the narrow link equals the IP-layer path capacity.

VIII.2.4 IP-layer used link capacity

For a given population of interest, the IP-layer used link capacity is

$$U(t,\Delta t) = \frac{n(t,\Delta t)}{\Delta t}$$

where n is the actual number of IP-layer bits transmitted between two measurement points separated by an exchange link generating successful IP packet transfer outcomes at the egress measurement point during a specified time interval $[t, t + \Delta t]$.

VIII.2.5 IP-layer link utilization

For a given population of interest, the IP-layer link utilization $V(t, \Delta t)$ of an exchange link is defined as the ratio between the IP-layer used link capacity $U(t, \Delta t)$ and the IP-layer link capacity $C(t, \Delta t)$. That is

$$V(t,\Delta t) = U(t,\Delta t)/C(t,\Delta t)$$

VIII.2.6 IP-layer available link capacity

For a given population of interest, the IP-layer available link capacity, $A(t, \Delta t)$, of an exchange link is the unused portion of the IP-layer link capacity during a time interval $[t, t + \Delta t]$. This can be calculated as the difference between the IP-layer link capacity and the IP-layer used link capacity. That is,

$$A(t,\Delta t) = C(t,\Delta t) - U(t,\Delta t)$$

or, equivalently

$$A(t,\Delta t) = C(t,\Delta t) * (1 - V(t,\Delta t))$$

VIII.2.7 IP-layer available path capacity

The definition of IP-layer available link capacity can be extended to a path. For a given population of interest, the IP-layer available path capacity during a specified time interval $[t, t + \Delta t]$ is defined as the smallest IP-layer available link capacity along that path. That is,

$$A_P(t,\Delta t) = \min_{i=1..n} A_i(t,\Delta t)$$

where A_i is the IP-layer available link capacity of the exchange link number i (i=1..n) on the path between two measurement points.

VIII.2.8 IP-layer tight link capacity

The IP-layer tight link is defined as the exchange link with the smallest IP-layer available link capacity of a path.

For a given population of interest, the IP-layer tight link capacity

$$C_{TL} = C_{TL}(t, \Delta t)$$

along a path is the IP-layer link capacity of the IP-layer tight link.

Note that the IP-layer available link capacity of the IP-layer tight link equals the IP-layer available path capacity. Observe also that the IP-layer tight link does not necessarily have to be the same exchange link as the IP-layer narrow link of a path.

VIII.2.9 Min, max and variance for IP-layer capacity framework parameters

The IP-layer capacity framework parameters represent mean values over a time interval $[t, t + \Delta t]$. Each such time interval can be divided into n subintervals $[t_1, t_1 + \Delta t / n]$, $[t_2, t_2 + \Delta t / n]$, …, $[t_n, t_n + \Delta t / n]$, each interval of length $\Delta t / n$, where $t_i = t + (i–1) \Delta t / n$. Note that $t_i + \Delta t / n = t_{i+1}$. That is, n non-overlapping equally sized subintervals are created. These subintervals are used to calculate the min, max and variance for the parameter mean value of interest.

The min function $m(t, \Delta t, n)$ is defined as

$$m(t,\Delta t,n) = \min_{i=1..n} P(t_i,\Delta t/n)$$

The max function $M(t, \Delta t, n)$ is defined as

$$M(t,\Delta t,n) = \max_{i=1..n} P(t_i,\Delta t/n)$$

The variance function $S(t, \Delta t, n)$ is defined as

$$S(t,\Delta t,n) = \frac{1}{n}\sum_{i=1..n}(P(t_i,\Delta t/n) - P(t,\Delta t))^2$$

where P corresponds to any of the following: IP-layer link capacity, IP-layer used link capacity, IP-layer available link capacity, IP-layer path capacity, IP-layer available path capacity or the IP-layer tight link capacity.

This clause is for further study. One question to be answered is whether the statistics should be calculated based upon dividing the time interval Δt (i.e., n intervals of length $\Delta t / n$) or based on several consecutive intervals (i.e., n intervals of length Δt).

VIII.3 Relation to [b-IETF RFC 5136]

[b-IETF RFC 5136] provides a discussion on terminology, mainly whether to use capacity or bandwidth for describing IP link and path characteristics. [b-IETF RFC 5136] proposes to use the term capacity. In order to harmonize with IETF, it is suggested that the same term be used in ITU-T. Note that the term available bandwidth is often used in the academic literature and corresponds to IP-layer available capacity. Further more, the term bandwidth in itself is often used to describe IP-layer link capacity.

The table below provides a mapping between the parameters that constitute the framework in this appendix and the definitions in [b-IETF RFC 5136]. The parameters are essentially the same; the IP-layer capacity framework also includes the IP-layer tight link capacity and the IP-layer capacity variation parameters.

ITU-T IP-layer capacity framework	[b-IETF RFC 5136]
IP-layer bits transferred	IP-layer bits
IP-layer link capacity	IP-type-P Link Capacity
IP-layer path capacity	IP-type-P Path Capacity
IP-layer used link capacity	IP-type-P Link Usage
IP-layer link utilization	IP-type-P Link Utilization
IP-layer available link capacity	IP-type-P Available Link Capacity
IP-layer available path capacity	IP-type-P Available Path Capacity
IP-layer tight link capacity	–
Min, max and variance for IP-layer capacity framework parameters	–

VIII.4 Use cases

The capability of measuring e.g., the IP-layer available path capacity between two hosts in an IP network is useful in several contexts. Examples include network monitoring, call-admission control and server selection. For example, measurement of IP-layer available path capacity in real-time opens up for adaptation based on that parameter directly (rather than measures such as loss or delay) in congestion control and streaming of audio and video.

In network monitoring utilizing active probing, both the IP-layer available path capacity and the IP-layer tight link capacity can be useful for characterizing the IP-layer tight link. The tight link is the "weakest link of the chain". Knowing the IP-layer available path capacity and the IP-layer tight link capacity, the utilization of the IP-layer tight link can be calculated. These parameters combined can for example help identifying common bottlenecks in a network.

For operators and for end users, it is important to be able to verify service level agreements (SLA).

Mobile broadband operators can buy transport services from other operators and thereby have a need to verify SLA parameters. Several of the parameters in the framework could be of interest.

M-Lab is one recent initiative by Google Inc, PlanetLab Consortium, New America Foundation's Open Technology Institute and researchers in the academic community. M-Lab provides end users with tools for SLA verification of their broadband connections. One of the parameters for SLA verification in this system is the "available bandwidth" which corresponds to the IP-layer available path capacity in this framework.

VIII.5 Items for further study

1) Regarding measurement point location within a host:
   a) Where should the measurement points be located within a host? How does the location of the measurement point affect the capacity values?
   b) Does fragmentation on sub-IP layers affect the capacity estimates? Can the location of the measurement point within a host compensate for sub-IP fragmentation?
2) The appendix does not explicitly address multipoint paths; however this is identified as an item for further study.
3) Is there a way of introducing a system for identification of the IP-layer tight link?
4) What additional information will help to harmonize this work with related efforts in e.g., IETF, ETSI, MEF, BBF and 3GPP?

5) For future methods of measurement, policing functions cause packet loss, and this form of limitation may require a different method of assessment from methods that rely on packet dispersion (e.g., BART).

Add the following standard to the bibliography:

[b-IETF RFC 5136] IETF RFC 5136 (2008), *Defining Network Capacity*.

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