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PERFORMANCE EVALUATION OF COMPUTER CAMPUS NETWORK SEGMENTS

The task of evaluating the performance of computer networks and their segments is considered - one of the main issues of the formation and functioning of modern campuses, educational spaces and corporate cloud solutions. To measure the characteristics of network performance, certain measurements were made on a sequence of packets arriving at some interface of a network device. Two types of existing measurements in the network are compared: active measurements and passive measurements. Conclusions are drawn about the possibility of measuring between any two nodes, or points, of the network based on preliminary data on the throughput capacity of individual network elements. The data transfer rate can be measured between any two nodes, or points, of a network, for example, between a client computer and a server, between the input and output ports of a router, and to analyze and configure the network to know the data on the throughput of individual network elements.

Keywords: *campus network ,rate of traffic ripple, ripple value, peak information rate, sustained information rate, variation in packet delay, round trip time.*

Introduction

The task of evaluating the performance of computer networks and their segments is one of the main issues of the formation and operation of modern campuses, educational spaces and corporate cloud solutions (1).

In order to evaluate a certain characteristic of network performance, it is necessary to carry out certain measurements on a sequence of packets arriving at some interface of a network device. We will compare two types of existing measurements in the network: active measurements and passive measurements(2).

Problem definition

Active measurements are based on the generation of special "measuring" packets in the source node. These packets must go through the network in the same way as packets whose characteristics we are going to evaluate. Measurements in the destination node are carried out on a sequence of "measuring" packages.

Figure 1 illustrates the idea of active measurements. The task is to measure the delays of packets of some application A, which are transmitted from the computer-client of application A to the computer-server of application A via the network.

Methods of solution

To measure packet delay, we will not use the stream generated by the client computer. Let us analyze two additional computers: a server-generator and a server-meter, additionally installed in the network. The server generator generates measurement packets, and the server meter measures the delays of these packets (3).

In order for the measured values to be close to the delay values of the packages of Appendix A, it is necessary that the measuring packets pass through the network along the same path as the packages of Appendix A, that is, you need to connect the generator server and the meter server as close as possible to the original nodes. In our example, this approximation is achieved by connecting additional nodes to the ports of the same switches S1 and S2 to which the original nodes are connected. In addition, it is necessary that the measuring packets have the structure of the original packets — dimensions and features placed in the packet headers. This is required in order for the network to serve them just like the original packages.

Measuring packages should not be generated too often, otherwise the network load may change significantly, and the measurement results will differ from those that would have been obtained in the absence of measuring packages. In other words, measurements should not change network conditions. Typically, the intensity of the generation of measuring packets does not exceed 20–50 packets per second. There is a special software that generates measuring packets and measures their characteristics upon arrival at the meter server.

Note that the placement of additional equipment, the creation of conditions for measuring packages that are close to the processing conditions of original packages without changing the network load, simplifies the measurement process and allows for their high accuracy. Since the generator server creates measurement packets, it can use a special packet format in order to place information necessary for measurement in them, for example, the time stamp of the package. Then, the meter server uses this time stamp to calculate the delay times. Obviously, in order for the delay measurements to be accurate, you need good synchronization between the generator server and the meter server. Since they are dedicated nodes in the active measurement scheme, such synchronization is easier to achieve than in the case of synchronization between the client and the application server A. In addition, sometimes measurement engineers do not have access to computers running

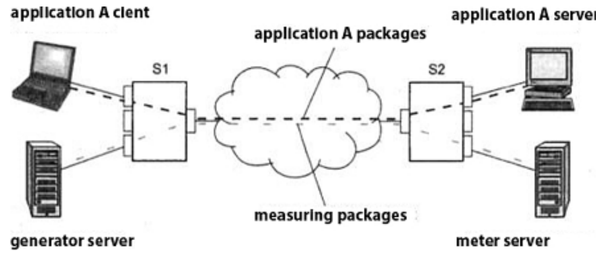

Fig. 1. Diagram of active measurements

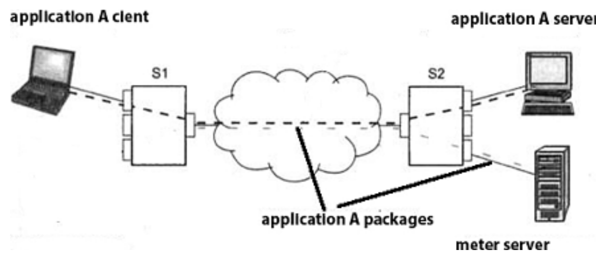

Fig. 2. Diagram of passive measurements

applications to install software for required measurements of incoming packets. And if such access exists, the client and server operating systems and the hardware platform are not optimized for accurate measurements of time intervals, which means they introduce large distortions in the results. However, the benefits of an active measurement scheme are not absolute. In some situations, the scheme of passive measurements is preferable (4).

Passive measurements are based on measurements of the characteristics of real traffic. This diagram is illustrated in Figure 2.

Arguments in favor of the active measurement scheme describe the problems that have to be solved when using the passive measurement scheme:

- the complexity of synchronizing client and server;
- additional and uncertain delays introduced by the universal multi-program operating systems of these computers;
- the absence in the header of the packet used by the application for transferring the time stamp over the network.

Some of these problems are solved by using a separate meter server. This server accepts the same packet input stream as one of the nodes participating in the packet exchange, whose characteristics need to be measured (the figure shows the case when the metering server is installed parallel to the application server A). In order for the meter server to receive the same input packet stream as the original node, they usually resort to duplicating the measured traffic to the port to which the meter server is connected. This feature, called port mirroring, is supported by many local area network switches. The meter server can operate under the control of a specialized operating system that is optimized to perform accurate measurements of time intervals.

Harder to solve sync problem. Some protocols carry time stamps in their service fields, so if, for example, application A uses such a protocol, then part of the problem is solved. However, in this case, the question remains open about the accuracy of the system time in the client's computer of Application A. Therefore, in passive mode, those characteristics that do not require synchronization of the transmitter and receiver are measured, for example, the proportion of lost packets is estimated.

A possible variant of the passive measurement scheme is the absence of a dedicated server meter. Some applications themselves measure the latency of incoming packets, for example, many applications of IP-telephony and video conferencing have such functions, since the information on packet delays helps to determine the possible reason for the unsatisfactory quality of the application.

Packet delay characteristics

The unit value of this metric is described as the transmission time of a certain type of packet between some two network nodes. A certain type is a package that has a certain set of predefined attributes; The standard does not strictly stipulate these features, but indicates that they may be, for example, the packet size, the type of application that generated the packet, the type of transport layer protocol that delivered the packet, as well as some others. The meaning of the feature set used is to isolate from the general packet flow arriving at the destination node those packets whose characteristics interest the measuring technician.

Since this definition takes into account the packet buffering time by the receiving node, the delay depends on the packet size. To obtain comparable results, it is desirable in the definition of the type of packages to set a specific packet size. RFC 2679 does not explain why the definition of delay was chosen depending on the size of the packet, but we can assume that this is due to the convenience of measuring the arrival time of the packet, since it can be measured by software only after the entire packet has been written to the operating system buffer. In practice, it is also impossible to understand whether a packet is of the correct type, when only its first bit is received(5).

In that case, if the packet did not arrive at the destination for some sufficiently long time, then the packet is considered lost, and its delay is undefined (it can be considered equal to infinity).

The sequence of measurements is recommended to be performed at random times, following the Poisson distribution. This order of measurement timing makes it possible to avoid possible synchronization of measurements with any periodic fluctuations in the network behavior, since such synchronization can significantly distort the observed picture.

RFC 2679 recommends using the following statistical estimates for one-way latency:

- quantile for a certain percentage, while the value of the percentage itself is not specified;
- average delay;
- minimum delay value (in the sample).

Quantiles are convenient for estimating delays in cases where the percentage of packet loss is high enough, so calculating the average value of the delay causes certain difficulties (you can ignore packet loss, but then we get too low an estimate). To calculate the quantile, the lost packets can be considered as packets arriving with an infinitely long delay, which, naturally, is greater than the quantile value.

Network response time is an integral characteristic of network performance from the user's point of view. It is this characteristic that the user has in mind when he says: "Today the network is slow."

The network response time can be represented as several items, for example, as in Figure 3:

- the time of preparation of requests on the client computer ($t_{client1}$),
- the time of the transfer of requests between the client and the server via the network ($t_{network}$),

- request processing time on the server (t_{server})
- response time from the server to the client via the network ($t_{network}$ again)
- processing time of responses received from the server on the client computer ($t_{client}2$).

Network response time characterizes the network as a whole, including the quality of the hardware and software of the servers. In order to separately assess the transport capabilities of the network, another characteristic is used — the Round Trip Time on the network.

Round Trip Time, RTT is among the IPPM standards. The turnaround time is a component of the network response time — this is the "pure" data transport time from the sender node to the destination node and back, without taking into account the time spent by the destination node for preparing the response:

$$RTT = 2 \text{ x } t_{net\ work}$$

The unit value of RTT is defined as the time interval between the sending of the first bit of a packet of a certain type by the sending node to the receiving node and the receipt of the last bit of this package by the sending node after the package was received by the node by the receiver and sent back.

In this case, the receiving node must send the packet to the sending node as quickly as possible so as not to distort due to the processing time of the packet. RFC 2861 recommends the same sequence of measurements of the turnover time as for a one-way delay, that is, random intervals obeying the Poisson distribution(6).

RTT is a convenient measurement characteristic, since receiving it does not require synchronization of the sending node and the receiving node (the sending node puts a time stamp on the packet being sent, and then upon arrival from the receiving node compares this mark with its current system time).

However, the informativeness of the turnover time is less than one-way delay, since information about the delay in each direction is lost, and this may make it difficult to find the problem path in the network.

Variation in packet delay, also called jitter, is very important for some applications. So, when playing a video clip, the delay itself is not very significant, for example, if all packets are delayed exactly ten seconds, the playback quality will not suffer, and the fact that the picture appears a little later than the server sent it will not even be noticed by the server (however, in interactive video applications such as video conferencing, such a delay will, of course, be noticeably annoying). But if the delays constantly vary from zero to 10 seconds, the quality of the clip playback will noticeably deteriorate, to compensate for such variable delays, you need to pre-buffer incoming packets for a time longer than the delay variation.

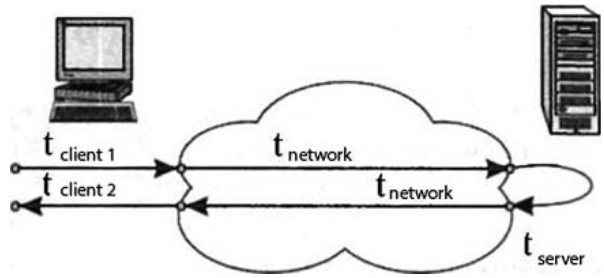

Fig. 3. Reaction time and Round Trip Time

The unit value of the delay variation estimate is defined in RFC 3393 as the difference of one-way delays for a pair of packets of a given type obtained in the measurement interval T(7).

As well as for one-way delay, the packet type can be specified by any signs, however, for definiteness of measurement of the delay variation, the sizes of both packets of the pair should be the same. The main question in this definition is how to choose a pair of packets on the measurement interval T? To answer this question, RFC 3393 introduces an additional function - the so-called selective function, which defines the rules for choosing a pair of packets. The standard does not define the exact meaning of this function, it only says that it must exist, and gives examples of possible functions. For example, pairs can be formed from all consecutive packets received in the interval. Another example is the selection of packets with specific numbers in the sequence of received packets.

To estimate the delay variation in accordance with the recommendations of RFC 3393, it is necessary to measure delays of certain pairs of packets. At the same time, another approach is often used to determine the variation of the delay, requiring only knowledge of the sample of one-way delays without grouping them into pairs that meet certain conditions. For example, in the document "IP Performance Metrics for Users", a delay spread is proposed as an estimate of the delay variation. The delay spread is defined as the difference between 75% and 25% one-way delay quantiles. Thus, in order to estimate the variation of the delay according to this definition, it is sufficient to obtain a sample of the values of one-way delay, and then find the corresponding quantiles.

Transmission Rate Characteristics

The data transfer rate (information rate) is measured over a period of time as a quotient from dividing the amount of data transmitted during this period by the duration of the period. Thus, this characteristic is always the average data transfer rate.

However, depending on the size of the interval in which the speed is measured, for this characteristic one of two names is traditionally used: average or peak speed.

Sustained Information Rate (***SIR***) is determined over a sufficiently long period of time. This is a medium-term characteristic, the period of time should be sufficient so that we can talk about the steady behavior of such a random variable, which is speed.

A period of control of this value must be specified, for example, 10 seconds. This means that every 10 seconds the speed of the information flow is calculated and compared with the requirement for this value. If such control measurements were not carried out, this would deprive the user of the opportunity to make claims to the supplier in some conflict situations. For example, if the supplier on one of the days of the month does not transmit user traffic at all, and on the remaining days permits the user to exceed the stipulated limit, the average speed per month will be normal. In this situation, only regular speed control will help the user to defend their rights.

Peak Information Rate (PIR) is the highest rate that a user stream is allowed to reach within a specified short period of time T.

This period is usually called the period of pulsation. Obviously, with the transfer of traffic, you can talk about this value only with some degree of probability. For example, the requirement for this characteristic can be formulated as follows: "The speed of information should not exceed 2 Mbit / s over a period of 10 ms with a probability of 0.95." Often, the probability value is omitted, implying its proximity to unity. Peak speed is a short-term characteristic. PIR allows you to assess the ability of the network to cope with peak loads characteristic of pulsating traffic and leading to congestion. If both speeds are specified in the SLA (SIR and PIR), it is obvious that pulsation periods should be accompanied by periods of relative "calm" when the speed falls below the average. Otherwise, the average speed will not be met.

The ripple value (usually referred to as B) is used to estimate the buffer capacity of the switch required to store data during an overload. The ripple value is equal to the total amount of data arriving at the switch during the allowed T interval (ripple period) of data transmission at peak rate (PIR):

$$B = PIR \times T$$

Another characteristic of the transmission rate **is the rate of traffic ripple** — the ratio of the maximum rate for a short period of time to the average rate of traffic measured over a long period of time. The uncertainty of time periods makes the ripple coefficient a qualitative characteristic of traffic.

Conclusion

The data transfer rate can be measured between any two nodes, or points, of a network, for example, between a client computer and a server, between the input and output ports of a router, and to analyze and configure the network to know the data on the throughput of individual network elements(8).
With the consistent nature of data transmission by different network elements, the total bandwidth of any composite path in the network will be equal to the minimum bandwidth capacity of the constituent elements of the route.

To increase the capacity of the composite path, you must first pay attention to the slowest elements, called bottlenecks.

The list of parameters and methods for their analysis represent a general assessment of the performance of segments in campus computer networks.

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ОЦЕНКА ПРОИЗВОДИТЕЛЬНОСТИ СЕГМЕНТОВ КОМПЬЮТЕРНЫХ КАМПУСНЫХ СЕТЕЙ

Введение. Задача оценки производительности компьютерных сетей и их сегментов есть одной из основных проблем формирования и функционирования современных кампусов, образовательных пространств и корпоративных облачных решений. Для оценки производительности сегментов кампусной сети на последовательности пакетов, поступающих на некоторый интерфейс сетевого устройства, рассматриваются два типа измерений в сети — активные и пассивные.

Цель — разработка процедуры оценивания для активных измерений — на основании генерации в узлеисточнике специальных «измерительных» пакетов, для пассивных — на основании способов перехвата характеристик реального трафика.

Методы. При разработке этой процедуры для построения оценочных алгоритмов использован набор процедур, рекомендованных *RFC* соответствующей тематики в сочетании со стандартными характеристиками структурного анализа сетевых инфраструктур.

Результаты. Разработана методика анализа сетевых потоков в кампусных сегментах. Сделаны выводы о возможности измерений между любыми двумя узлами, или точками сети на основании предварительных данных о пропускной способности отдельных элементов сети.

Заключение. Список параметров и методов их анализа представляет собой общую оценку эффективности сегментов в компьютерных сетях. При этом для анализа и настройки сети необходимо знать данные о пропускной способности отдельных элементов сети. Поскольку последовательный характер передачи данных различными элементами сети формирует общую пропускную способность любого составного пути в сети, равную минимальной пропускной способности составляющих элементов маршрута, то для повышения пропускной способности составного пути необходимо, в первую очередь, ускорять самые медленные элементы.

Ключевые слова: кампусная сеть, скорость пульсации трафика, величина пульсаций, скорость пиковой информации, постоянная скорость передачи данных, время задержки пакета, диаметр компьютерной сети.

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ОЦІНКА ПРОДУКТИВНОСТІ СЕГМЕНТІВ КОМП'ЮТЕРНИХ КАМПУСНИХ МЕРЕЖ

Вступ. *З*адача оцінювання продуктивності комп'ютерних мереж та їх сегментів є однією з основних проблем формування та функціонування сучасних кампусів, освітніх просторів і корпоративних хмарних рішень. Для оцінювання продуктивності сегментів кампусової мережі на послідовності пакетів, що надходять на деякий інтерфейс мережевого пристрою, розглядаються два типи вимірювань в мережі — активні та пасивні.

Мета. Метою даної статті є розробка процедури оцінювання. Для активних вимірювань — на підставі генерації в джерельному вузлі спеціальних «вимірювальних» пакетів, для пасивних — на підставі способів перехоплення характеристик реального трафіку.

Методи. При розробці цієї процедури для побудови оціночних алгоритмів використано набір процедур, рекомендованих *RFC* відповідної тематики у поєднанні зі стандартними характеристиками структурного аналізу мережевих інфраструктур.

Результати. Розроблено методику аналізу мережевих потоків у кампусних сегментах. Зроблено висновки про можливість вимірювань між будь-якими двома вузлами, або точками мережі на підставі попередніх даних про пропускну здатність окремих елементів мережі.

Висновки. Список параметрів і методів їх аналізу являє собою загальну оцінку ефективності сегментів в комп'ютерних мережах. При таких умовах, для аналізу і настроювання мережі необхідно знати дані про пропускну здатність окремих елементів мережі.

Оскільки послідовний характер передачі даних різними елементами мережі формує загальну пропускну здатність будь-якого складового шляху в мережі такою, що вона є мінімальною з пропускних спроможностей складових елементів маршруту, то для підвищення пропускної здатності складеного шляху необхідно, в першу чергу, прискорювати самі повільні елементи.

Ключові слова: кампусна мережа, швидкість пульсації трафіку, величина пульсацій, швидкість пікової інформації, постійна швидкість передачі даних, затримки пакета, час обороту пакета, діаметр комп'ютерної мережі.