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Loss and Delay Measurements of Internet Backbones

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Abstract

As the Internet evolves into a universal network for all communication needs, it has to stand up to the high quality standards of traditional networks, such as the telephone network for voice communications. Multimedia applications are particularly sensitive to various impairments introduced by IP networks, such as packet loss, delay and delay jitter. In this paper, we study loss and delay measurements taken over the Internet and we provide a detailed characterization thereof. We focus on wide-area backbone networks, which constitute an important part of long-distance communication. Our study is based on a rich data set that provides valuable insights into the behavior of Internet backbones today, and in particular into how they affect multimedia traffic. We find that most of the problems observed seem more related to reliability, network protocols and router operation rather than to traffic load and traditional quality-of-service issues. Furthermore, the characterization and modeling of packet loss, delay and delay jitter can be used by the research community as input to various problems related to the design and evaluation of network- and application-layer mechanisms.

Keywords: IP backbone networks; Loss; Delay and jitter; Measurements; Characterization and modeling

1. Introduction

As the Internet evolves into a ubiquitous network, it is important to understand its performance and capability of supporting various services at high quality. We are particularly interested in low-latency multimedia applications, such as Voice over IP (VoIP), video-conferencing and audio/video streaming. These applications are particularly demanding for two reasons: first, they have stringent requirements in terms of packet loss, end-to-end (e2e) delay and delay jitter; second, there are already high quality standards set by traditional networks, such as the telephone network for voice communications.

The first users of VoIP were eager to tolerate bad quality because it was a free service. However, as VoIP evolves, it needs to achieve the high quality of traditional telephony. Simply stated, the problems that occur in the Internet and can affect the quality of voice and video communication are packet loss, delay and delay jitter. Loss and delay jitter can be due to congestion in the network, leading to packets getting dropped in the routers, or failure of network components leading to a reconfiguration of the network. Here the issue is how extensive are loss, delay and jitter, how bad are their effects, and whether they can be concealed at the destination.

The contribution of this paper is the collection and characterization of loss and delay measurements over a representative set of Internet backbone paths. We provide valuable insights into the behavior of Internet backbones, in particular with respect to their ability to support multimedia traffic. Furthermore, we provide a detailed characterization, and when possible modeling, of loss, delay and delay jitter that can be used by the research community to capture the backbone network behavior.

Our study is based on a rich data set that was collected by RouteScience Technologies Inc. Probes were sent between five facilities, over a large number of different paths (43 paths belonging to seven different Internet providers in the continental US), every 10 ms for a continuous period of 2.5 days, and accurately time-stamped using GPS. The study of this data set reveals a wide range of behavior among providers: while some backbone networks exhibit excellent behavior, some other have consistent problems that severely impair the performance of multimedia traffic. Furthermore,

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the problems that we identify seem more related to reliability, network protocols and router operation, rather than to traffic load and traditional Quality of Service (QoS) issues.

We focus on wide-area backbone networks, for which we have extensive data available. These are an important part of the end-to-end path for all long distance communications, including VoIP calls that are serviced by a combination of a switched telephone network in the local area and the Internet for the long haul. Problems on the backbones will be experienced by all such calls; therefore, they need to be well understood and fixed, regardless of what takes place elsewhere in the path.

This study takes a multimedia perspective in the following sense. First in the collection phase, we sent active probes emulating voice and video traffic. Then in the characterization phase, we analyzed properties such as packet loss, delay and delay jitter, which are of critical importance to interactive or streaming multimedia (as opposed to average round-trip times that would be of interest to TCP). This way, we draw conclusions about the capability of the studied networks to support multimedia applications. Furthermore, one can use our statistical characterization to capture the behavior of these networks and evaluate adaptive mechanisms at the end-systems, such as playout scheduling, multipath streaming or rate-distortion optimized streaming.

The outline of the paper is as follows. In Section 2, we review work related to network measurements and multimedia quality. In Section 3, we describe the measurement setup and collection. In Sections 4 and 5, we describe the loss and delay characteristics, respectively, observed in the measurements; we provide representative examples and a detailed statistical characterization. In Section 6, we briefly discuss the effect of the observed network impairments, their possible causes and remedies. Section 7 concludes the paper.

2. Related work

There has been an extensive amount of work on measurements and characterization of the Internet. Different studies take a different perspective depending on their specific interest (e.g. part of the Internet hierarchy under study, network protocols designed or evaluated, applications and performance metrics of interest) as well as on implementation constraints.

We are interested in the quality of multimedia communications over the public Internet. The following studies had the same objective. In [1], the delay and loss experienced by audio traffic was measured; they found that the delay variability had the form of spikes and modeled it as the result of multiplexing audio and interfering traffic into a single queue. In [11,18], audio traffic was also studied over the MBONE, and loss rates, burstiness and correlation between loss and delay were characterized. In [11], delay variability was found to have the form of spikes and playout scheduling algorithms were proposed to deal with these spikes. In [6], a large-scale experiment was conducted, where low-rate MPEG-4 video was streamed to a large number of clients and cities, and statistics for the quality of the sessions were provided. The study of loss and delay in [12], turned out to be heavily used today in the video community, particularly for modeling network delay using Gamma distributions. Poisson flows were used in [19] to sample the network, and the constancy of delay and loss on Internet paths was studied. Finally, [4] developed a measurement technique for inferring the state and performance of TCP-based applications based on passive measurements.

The topic of measurements from the edge of the network is important in far too many contexts to be surveyed here exhaustively. For example, a tool for inferring ISP topologies and various metrics of interest based on measurements from the edge was developed in [16]. User-level Internet-path diagnosis was provided in [10]. In general, being able to 'measure the black box' is important for applications to optimize their performance.

We focus on backbone networks in the continental US, which are in general sufficiently provisioned, so they are typically believed not to introduce any impairment. Indeed, in our study we observed delay and loss patterns on those networks that seem mostly related to the network and router operation, rather than to traffic load and congestion. Similar patterns had also been observed on backbone networks in [14,15]. [5] Investigated the stability and the failures of wide-area backbones due to the underlying switching system as well as due to the software and hardware components specific to the Internet's packet-switched forwarding and routing architecture. Recent studies of the Sprint's backbone network, [9], focused on link failures and their impact on voice traffic. They also studied the delay caused by a backbone router and identified periods during which the routers were take from serving packets [13].

A preliminary version of this paper appeared in [17]. This journal paper is significantly extended by additional materials, i.e. a complete and detailed classification and characterization of the measurements based on the work in [7]. Finally, in our previous work [8], we focused on the VoIP quality, we developed a methodology for mapping network parameters to voice subjective quality, we simulated voice calls and provided statistics on their quality. In contrast to [8], this paper focuses on the measurements themselves and on characterizing, and when possible modeling, the loss, delay and jitter observed therein. This characterization can be used by other researchers as input to problems related to the design and evaluation of network- and application-layer mechanisms.

3. Measurement set

Our study is based on measurements provided by RouteScience Technologies Inc. Facilities have been installed in five major US cities: San Jose in California (SJC), Ashburn in Virginia (ASH), Newark in New Jersey (EWR), Thornton in Colorado (THR) and Andover in Massachusetts (AND). These