

Regular Paper

Spread of IP Anycast and GSLB

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Abstract: Content Delivery Networks (CDN) are regarded as an important internet infrastructure that supports business on the Internet. Network latency has been identified as an important metric to improve the quality of service (QoS) of CDNs. Given the limitations to response times because of the geographic distance between servers and end-users, CDN providers have developed architectures such as Global Server Load Balancing (GSLB) and IP Anycast to realize fast responses. This study investigates how GSLB and IP Anycast contribute towards improving QoS from the users' perspective. User traffic in the campus network was analyzed and it was found that: 1) 93.8% of the traffic reaps the benefits of well operated internet services, i.e., they have a Round-Trip Time (RTT) of less than 0.1 second. 2) Although GSLB is still the primary architecture for realizing fast responses, IP Anycast supports 5.9% of the traffic.

Keywords: IP Anycast, GSLB, CDN, RTT

1. Introduction

In a survey about e-commerce websites, the BBC reported that shoppers are likely to abandon a website if it takes longer than four seconds to load [1]. In another study, Nielsen claimed that a page response of 0.1 seconds gives the feeling of instantaneous response [2]. Here, Content Delivery Network (CDN) is regarded as a critical infrastructure that supports business on the Internet. Consequently, CDN providers have developed architectures to realize fast responses, i.e., lower latency.

Although the network latency is considered an important aspect of Quality of Service (QoS), there are limitations in terms of reducing latency because of the size of the earth and the speed of light. Therefore, we cannot realize a response time of less than 0.1 second if, for example, the server is located in North America and its users access the server from Asia. In other words, facilitating the exchange of data packets over the Pacific Ocean in less than 0.1 second is not feasible.

Therefore, researchers have developed architectures such as Global Server Load Balancing (GSLB) [3] and IP Anycast [4] to realize fast responses. GSLB uses a DNS mechanism to find the IP address of the nearest server that offers the same service, for example, web content with the same URL. IP Anycast is an IP layer mechanism that provides similar functionality. Although there are several studies on the characteristics of their architectures, there is a need to study and evaluate their QoS from a user perspective. In this work, the authors report on how GSLB and IP Anycast contribute towards improving QoS from a user perspective. In particular, this study reports the measurement results of Round-Trip Time (RTT) for GSLB and IP Anycast. Through

comparisons of RTTs from sites in Asia, North America, and Europe, this study attempts to analyze how GSLB and IP Anycast reduce RTTs.

The remainder of the paper is structured as follows. First, Section 2 discusses the seminal works in this domain. Section 3 presents an overview of the measurement experiments and Section 4 reports the results of the experiments. Section 5 presents the discussion and findings of this work. Finally, Section 6 presents the conclusions.

2. Related Works

Although network latency is not the only factor that determines the user's perception of QoS, e.g., Ref. [5], it is still regarded as a critical issue in improving Quality of Service (QoS) [6] and [7]. This study regards RTT as a direct index for response time; hence, this study attempts to investigate the situation in the real world.

An important approach to reducing RTT is the improvement of communication protocol. For example, Quick UDP Internet Connections (QUIC) [8] is an approach to reducing the response time by using a new communication protocol, wherein it eliminates the process required for TCP and TLS handshake [9]. Some internet browsers such as Chrome employ QUIC to reduce the response time. However, as mentioned earlier, there are physical limitations to reducing RTT given the distance to be traversed by the data packet. Therefore, without a sophisticated mechanism to control traffic routes, a RTT of less than 0.1 second cannot be realized. QUIC alone cannot realize the required reduction in RTT.

Architectures such as GSLB [3] and IP Anycast [4], [10], [11] were developed to realize a traffic control wherein data transfer took place between the nearest sites. These architectures attempt to find the best/nearest server for each user. In particular, IP Anycast has been used for stateless service, such as DNS on UDP, because it requires a stable route operation [12]. However, the

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stable route and Border Gateway Protocol (BGP) operation [13] of today has enabled the use of IP Anycast for stateful service such as TCP. For example, Refs. [14], [15], [16] have reported the deployment of IP Anycast. These studies analyze various characteristics of IP Anycast, such as route appropriateness, from the service provider's perspective. Therefore, there exists a need to study QoS from the users' perspective. This paper reports how GSLB and IP Anycast contribute towards improving QoS from the users' perspective.

RTT is the basic metric for measuring latency. This study utilizes RTT to analyze the use of GSLB and IP Anycast (See next Section for details). Given the importance of measuring RTT, there have been significant efforts to measure RTT, such as Refs. [17] and [18]. These efforts report on the large-scale measurement results of network latency; in other words, they report RTT data from various measurement points. However, in this study, we use data from only three such sites. Through the selection of geographically distant sites, we attempt to investigate the effect of GSLB and IP Anycast on RTT.

Early results of this study are reported at the COMPSAC 2019 ADMNET Workshop^{*1}. The workshop paper is based on the data of a single day (Feb. 20, 2019). This paper uses the data from one week (16th–22nd of June 2019) to complete the analysis. Figures 1, 3–14, 16, 17 were redrawn based on the new data. Figures 15, 18–20 enhance the credibility of the results by providing new analyses.

3. Network Configuration

In this study, we used the traffic data between our campus network and the Internet. The outline of the campus network is shown in **Table 1** Site_A. In addition, **Fig. 1** shows the classification of destination hosts (precisely IP addresses) based on the Geo-IP data [19]. As shown in Fig. 1 (a), 14.1% of destination hosts are domestic hosts (inside Japan), 25.8% are North American hosts, and 12.9% are based in Europe. Asian (except Japanese) sites constitute 44.5% of the total number of sites, when classified based on Geo-IP. Furthermore, if the data is represented in terms of transferred volume, the ratio of Asian sites

decreases (Fig. 1 (b)). On the other hand, traffic corresponding to IP addresses of North America constitutes 51.9%, while that corresponding to Europe constitutes 4.5%. The share of domestic traffic (inside Japan) is 41.3%, while the remainder regions (Asia, South America, etc.) constitute 2.3%.

It should be noted that if some mechanism to reduce communication delay is not used, 56.4% of traffic (i.e., transferred volume between hosts in North America and Europe) cannot be reached within 0.1 second. Although Fig. 1 is based on the Geo-IP data and its inaccuracy is a well-known fact, e.g., Ref. [20], this Figure makes us start the study on the mechanism to reduce communication delay since we knew the short latency we have every day (See Section 4 for details). Here, the inaccuracy of Geo-IP data is not the issue of our research. We made the RTT based study to analyze the spread of IP Anycast and GSLB as the mechanism to reduce communication delay.

To analyze the latency of the traffic, we used the data log obtained from the local firewall and DNS server (**Fig. 2** Site_A). The firewall monitored all traffic between our campus network and the Internet. The firewall log contains information about access time, source and destination IP addresses, port numbers, protocol, number of packets, and volume of transferred data for each transaction. Here, the firewall determines the division of each transaction based on 1) the same 5 tuples (combination of protocol, source and destination IP addresses, and ports) in short interval, and 2) TCP SYN and FIN/RST packets. To elaborate on its mechanism, the firewall identifies transactions by the same 5 tuples. In

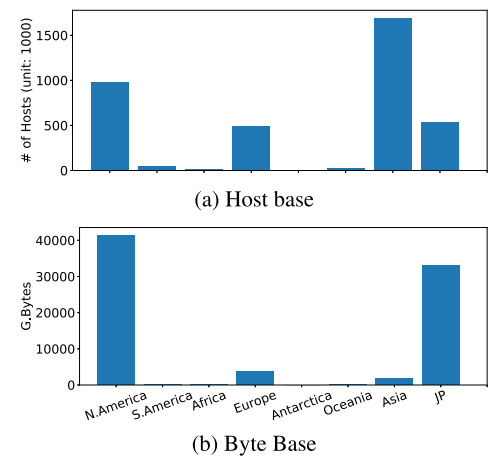


Fig. 1 % of each continents.

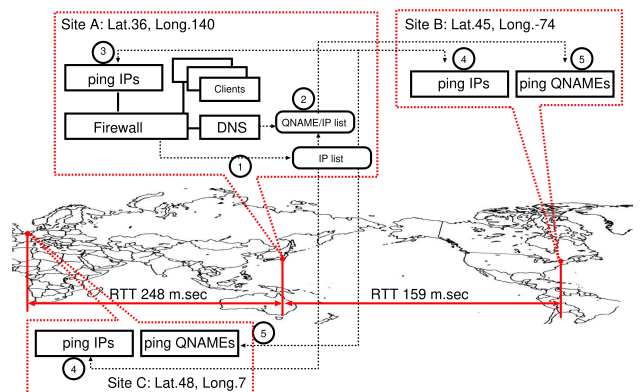


Fig. 2 Location of sites and their configurations.

Table 1 Outline of measured sites.

Site _A	
Latitude	36.109577
Longitude	140.101176
Date of IP traffic measurement	16–22 June 2019
Date of RTT measurement	17–23 June 2019
# of Clients	21,832
# of DST IPs	1,430,847
Transferred Data	65.625 Tera byte
Site _B	
Latitude	45.315082
Longitude	-73.877903
RTT from Site _A (14 Mar.2019)	159 millisecond
Site _C	
Latitude	48.573405
Longitude	7.752111
RTT from Site _A (14 Mar.2019)	248 millisecond

^{*1} Yoshida, K., Fujiwara, K., Sato, A. and Sannomiya, S. Spread of Any-cast and GSLB. In 43rd Annual Computer Software and Applications Conference (COMPSAC), Vol.2, pp.30–35, IEEE (2019).

the case of UDP, a transaction is considered to end after a certain interval. In the case of TCP, a new transaction is recognized by a TCP SYN packet, and TCP packets with the same 5-tuples are considered for the same transaction. The TCP connection is considered to end when a TCP FIN or RST packet is received or timed out.

The DNS server is the main full-service resolver of our campus network. It resolves queries from our campus networks. The server's log has information about access time, QNAME, and reply for the query (e.g., IP address and CNAME). Here, we only analyze DNS queries from our campus network. The DNS queries requesting the address of the Tsukuba servers from outside are excluded from the analysis. In other words, all host computers (whether they are www servers, mail servers, or www clients) act as DNS clients in this study.

We also setup Site_B and Site_C to measure latency from North America and Europe, respectively. We rent a VPS service^{*2} for this purpose. Given that Site_A is in Asia (Japan), the minimum possible RTT between Site_A and Site_B is 159 ms. The minimum RTT between Site_A and Site_C is 248 ms.

The following procedure was used to measure the latency of the traffic:

- Step 1:** Extract list of the destination IP addresses from firewall log
- Step 2:** Find QNAME for the destination IP address from DNS log
- Step 3:** Send five ICMP packets to each destination IP address from Site_A, and take minimum response time as RTT for the destination.
- Step 4:** Observe RTT for each destination IP address from Site_B and Site_C by using the same procedure.
- Step 5:** Observe RTT for each destination QNAME from Site_B and Site_C by using same procedure.

In Steps 3, 4 and 5, we send five ICMP packets and record the minimum response time to observe RTTs. We use minimum value as RTT because we focus on IP Anycast servers. To find IP Anycast servers, minimum latency is important to use physical limitation as the classification criteria. Accordingly, we observe the minimum RTT.

In Steps 3 and 4, packets are sent to the same destination IP address from geographically distant sites, i.e., from Site_A and Site_B. By finding total response time of less than 159 ms, we can find the use of IP Anycast. If host_x is a single physical entity located in Japan or North America, the sum of "RTT between Site_A and host_x" and "RTT between Site_B and host_x" will exceed 159 ms. To reduce this sum, two physical entities are required. In other words, the use of IP Anycast is necessary.

In Step 5, packets are sent to QNAME from Site_B and Site_C. QNAME requires GSLB to find the nearest IP address for the same service. Thus, the difference between Steps 4 and 5 enables the analysis of GSLB usage.

4. Analysis of University of Tsukuba Data

4.1 Step 3: Achieved QoS

Figure 3 shows the volume of traffic between Site_A and the Internet for 15-minute time periods. In the above figure, the solid line indicates the total volume of transferred data. "X" mark indicates the volume of traffic between Site_A and sites whose RTT from Site_A is less than 0.1 second. Similarly, the dotted line at the bottom of the figure indicates the volume of traffic between Site_A and sites whose RTT is more than 0.2 second. As shown in Fig. 3, most of the traffic exists between sites whose RTT is less than 0.1 second.

Figures 4 and 5 show the ratio of RTT more clearly. Figure 4 shows the fluctuation of RTT, and Fig. 5 shows the CDF (cumulative distribution). Although the ratio of RTT at 15-minute intervals appears to fluctuate, approximately 93.8% of the traffic is exchanged between Site_A and hosts whose RTT is less than 0.1 second.

As shown in Fig. 5, about 48.0% of hosts have an RTT of less than 0.1 second. If this data is weighted with traffic volume, we can find that approximately 93.8% of traffic has RTT less than 0.1 second.

4.2 Step 4: IP Anycast

To clarify the effect of IP Anycast, we first created Figs. 6 and 7. Figure 6 is the same as Fig. 5 except it uses RTT from Site_B,

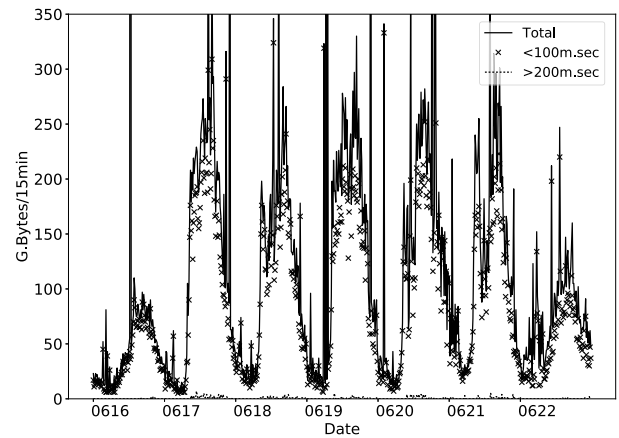


Fig. 3 G.byte per 15 min.

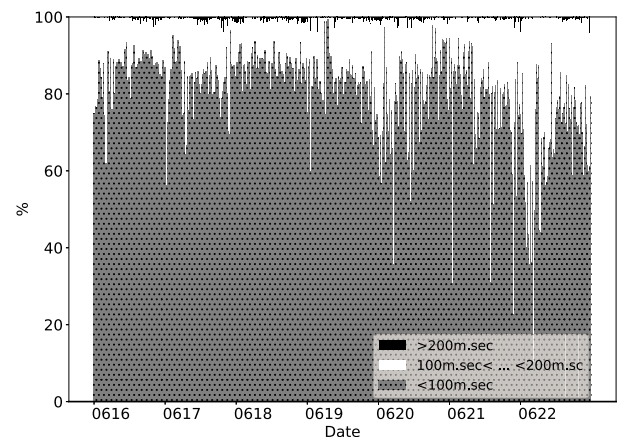


Fig. 4 Fluctuation of RTT ratio.

^{*2} <https://www.ovh.com/world/> (accessed 2019-07-05)

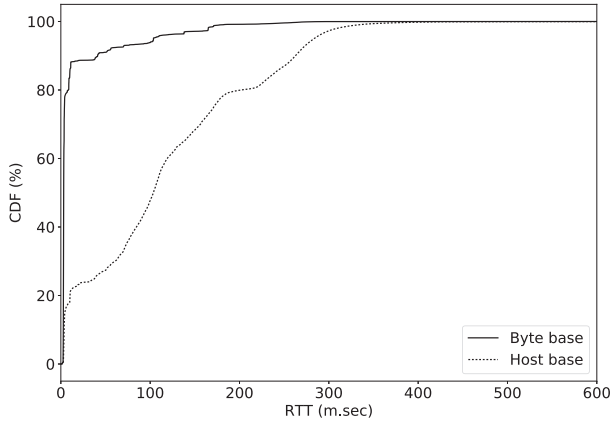


Fig. 5 Step 3: RTT from Site_A.

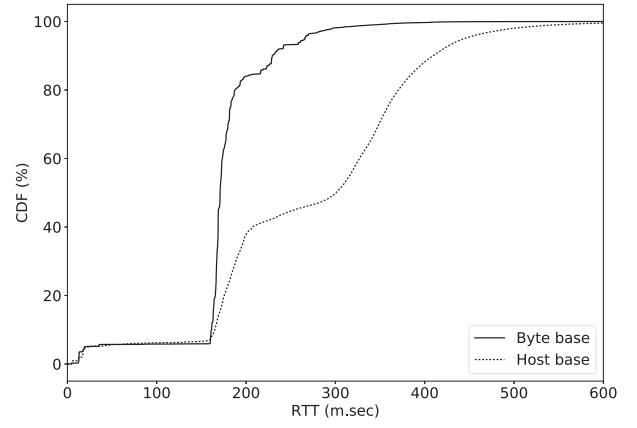


Fig. 8 Step 4: (RTT from Site_A)+(RTT from Site_B) (IP base).

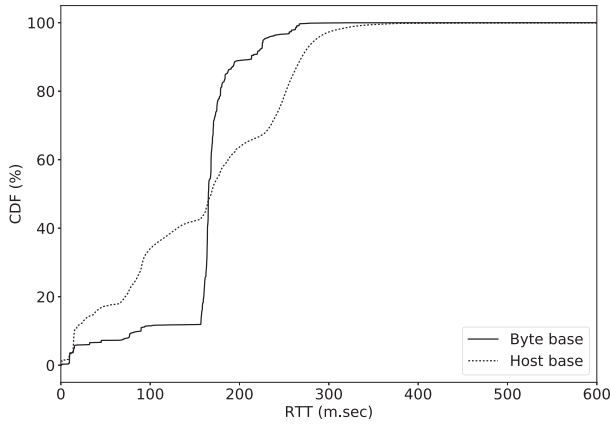


Fig. 6 Step 4: RTT from Site_B (IP base).

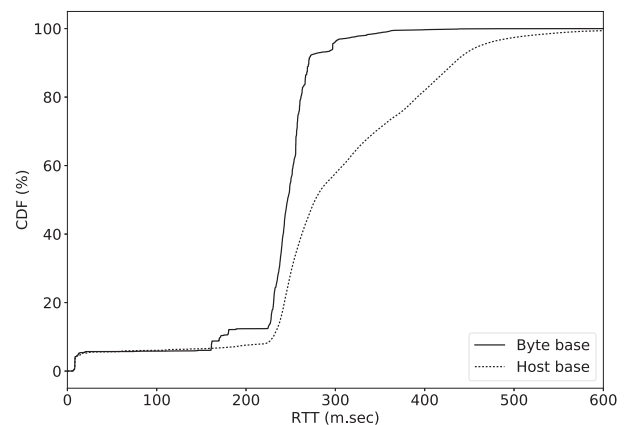


Fig. 9 Step 4: (RTT from Site_A)+(RTT from Site_C) (IP base).

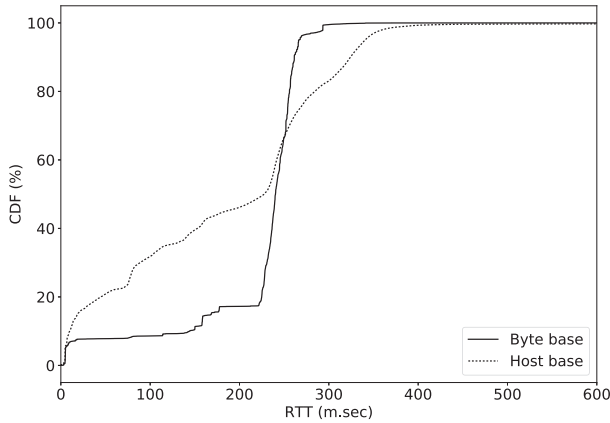


Fig. 7 Step 4: RTT from Site_C (IP base).

i.e., North America. Figure 7 uses RTT from Site_C, i.e., Europe.

Here, to measure RTT from Site_B and Site_C, the IP addresses observed at Site_A were used. The IP addresses and volume of data transferred, which were recorded in the firewall log, were used for this purpose. This is to emulate the situation where the accesses are to the same IP addresses from North America (Fig. 6) and Europe (Fig. 7).

As shown in Fig. 6, even if the users access the same IP addresses from North America, 11.5% of traffic (byte base, host base is 34.0%) has an RTT of less than 0.1 second. If the users access the same IP addresses from Europe, 8.6% of traffic (byte base, host base is 31.8%) has an RTT of less than 0.1 second. In other words, IP Anycast contributes the realization of access

latency of less than 0.1 second for 8.6–11.5% traffic.

Here, both 8.6% and 11.5% include the combination of IP Anycast traffic and domestic traffic (inside North America and Europe, respectively). To distinguish the traffic of IP Anycast from domestic traffic, Figs. 8 and 9 were generated. Figure 8 is the same as Fig. 5 except that the total RTT from Site_A and Site_B are used. Figure 9 uses the total RTT from Site_A and Site_C.

Here, the RTT between Site_A and Site_B is 159 ms. If the total RTT from Site_A and Site_B to some destination host is less than 159 ms, such destination host can be classified as a host served by IP Anycast. Based on the same criteria, such destination hosts can also be classified using the data from Site_C. To be precise, we use 140 ms as a criterion to distinguish the use of IP Anycast. Here, a fluctuation of 19 milliseconds is assumed in RTT measurement (See Section 5.1 for details).

As shown in Fig. 8, 5.9% of traffic (byte base, host base is 6.5%) has a total RTT of less than 140 ms. In other words, IP addresses for 5.9% of traffic can be reached in short RTTs both from North America and Japan. As shown in Fig. 9, IP addresses for 5.9% of traffic can be reached in short RTTs both from Europe and Japan. In addition, it can be concluded that IP Anycast is used to realize such IP addresses.

Based on these results, we conjecture that 5.9% of traffic uses IP Anycast at Site_A, as on 16–22 June 2019.

4.3 Step 5: GSLB

To accurately determine the effect of GSLB, we emulate the

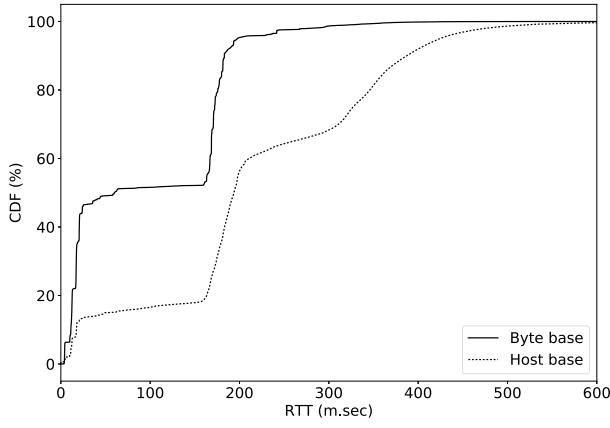


Fig. 10 Step 5: RTT from Site_A + Site_B (QNAME base).

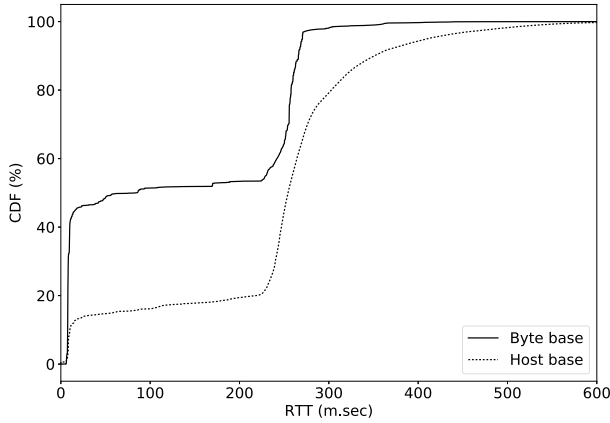


Fig. 11 Step 5: RTT from Site_A + Site_C (QNAME base).

situation where the accesses are to the same QNAME from North America (Fig. 10) and Europe (Fig. 11). For this purpose, the IP addresses are replaced in the firewall log with corresponding QNAME recorded in DNS log, and the RTT are measured based on the modified firewall log.

The use of QNAME enables the GSLB to find the nearest IP address for the given QNAME. Thus, the effect of GSLB can be evaluated.

Figure 10 is the same as Fig. 5 except we used RTTs from Site_A and Site_B, and Fig. 11 is the same as Fig. 5 except we used RTTs from Site_A and Site_C. In both figures, RTTs for QNAME are used to clarify the effect of GSLB.

From these figures, we can see that hosts for 52.1% of traffic (byte base, host base is 17.7%) can be reached within 140 ms from North America. Moreover, hosts for 51.8% of traffic (byte base, host base is 17.7%) can be reached within 140 ms from Europe. Given that 52.1% and 51.8% of traffic are assumed to be traffics of GSLB and IP Anycast from North America and Europe respectively. We can assume 46.2 (about 52.1–5.9)% of traffic is controlled by DNS based GSLB.

Considering the RTT shown in Fig. 6, about 6.0% (i.e., 11.9–5.9) of traffic is from North America. Also, about 3.7% (i.e., 9.6–5.9) of traffic is from Europe.

5. Discussion

5.1 Reliability of our Experiments

In this study, we use the ICMP response to measure the RTT.

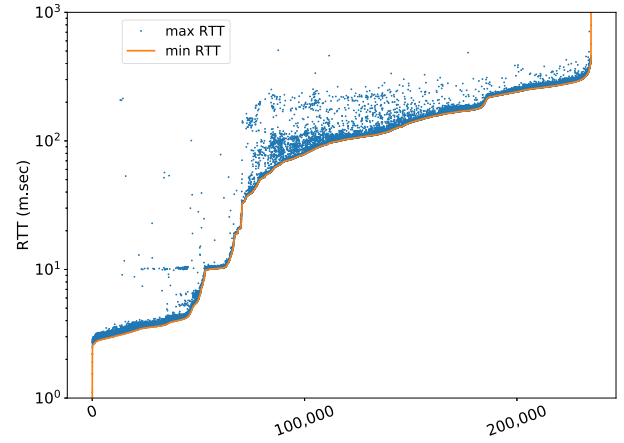


Fig. 12 Ping stability at Site_A.

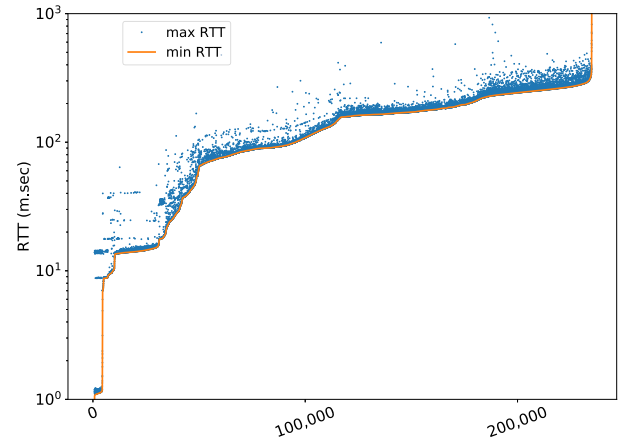


Fig. 13 Ping stability at Site_B.

Unlike the prior forecasts, many sites responded to ICMP. Although the number of hosts that responded to ICMP was approximately 60.0%, most of the sites with large transferred volume responded to ICMP. Thus, the RTT was observed for 85.0% of transferred data. Consequently, the byte base calculation in this study can be regarded as sufficiently reliable.

Another important factor that cements the reliability of this study is the results observed by ICMP response, i.e., ‘ping’ command results. To observe the reliability of the “ping” command results, we compared the maximum RTT and the minimum RTT at the same site. During June 17–23 2019, we observed the RTT for the accessed site a day ago and measured multiple RTTs for the same site. The results are shown in Figs. 12, 13, and 14.

In these figures, the lines indicate the minimum RTT of the experiment. Dots indicate the maximum RTT. As can be observed in the figures, there are slight differences. However, the correlation between maximum and minimum is high (0.966, 0.893, 0.905 for Figs. 12, 13, and 14, respectively).

Traffic congestion seems to create these differences. As shown in Fig. 15, 90.0% of the fluctuation (i.e., max - min) is less than 19 milliseconds. Thus, the results of RTT measurements appear to be reliable. And we use 19 milliseconds as the criteria to distinguish the use of IP Anycast.

Figure 16 presents an overview of the results of this study. As per our estimation, traffic of CDNs supported by IP Anycast accounts for 5.9% and traffic supported by GSLB accounts for

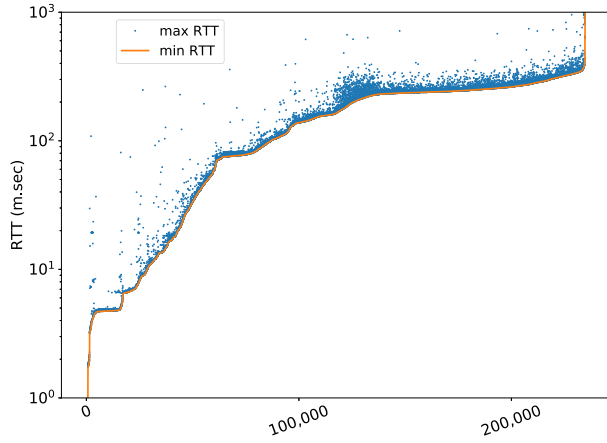
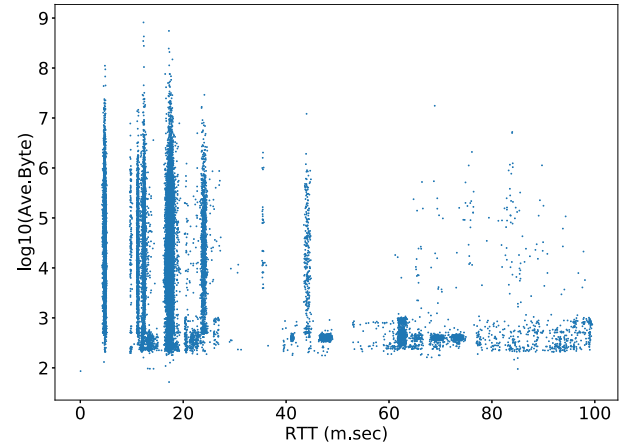
Fig. 14 Ping stability at Site_C.

Fig. 17 Average transferred bytes.

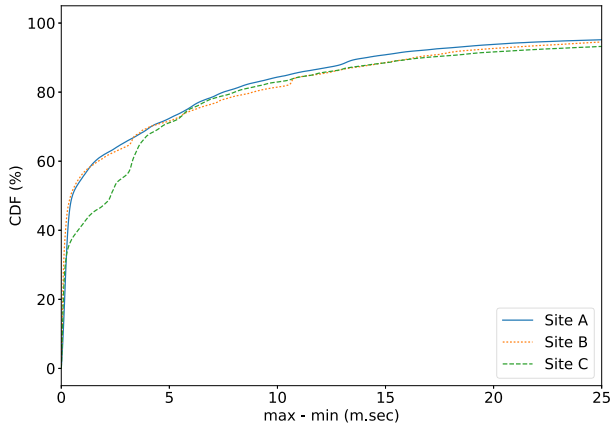


Fig. 15 Ping fluctuation.

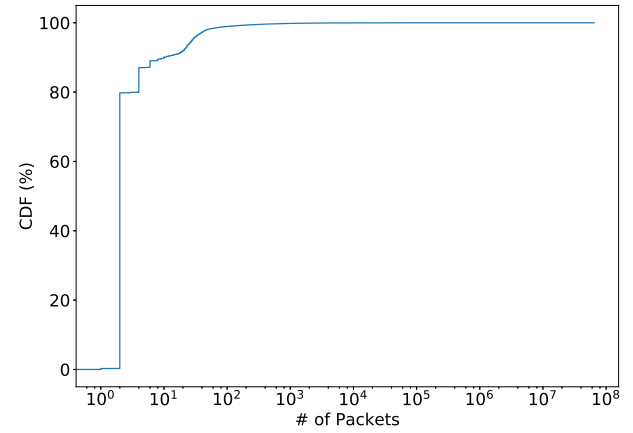


Fig. 18 # of packets in each IP Anycast transaction.

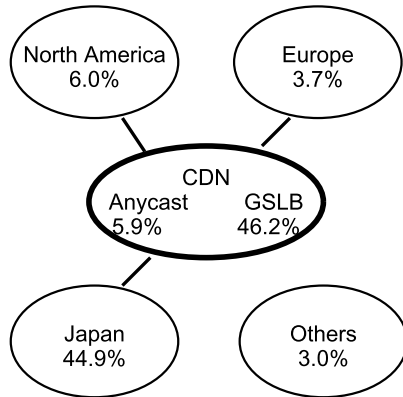


Fig. 16 Overview of results.

46.2%. Domestic traffic accounts for 44.9%. The sum total of these numbers is 106.7, which exceeds 100.

This inconsistency can be attributed to the short RTT between Site_B and Site_C (81 ms), which leads to some hosts being identified as domestic hosts both in North America and Europe. Given that ‘Others’ includes Asian sites whose RTT from Site_A are less than 0.1 second, such sites are accounted for twice.

It is evident that the 140 ms criteria, which were used to determine the use of IP Anycast and GSLB, may not be the ideal criteria to distinguish domestic hosts from others. In our experience, there are roughly three clusters of hosts, Asia, North America, and Europe. Within Asia, the RTT tends to be less than 50 ms. Thus, the criteria of 140 ms is useless when attempting to distin-

guish the traffic inside Asia from domestic traffic inside Japan. Although the minimum RTTs between Europe and North America are approximately 80 ms, RTTs inside North America sometimes exceed 80 ms. Thus, RTT base analysis is insufficient to analyze domestic traffic.

However, because of the geographical size of the Pacific Ocean, RTTs can be a good metric to determine if IP Anycast and GSLB are being utilized. The results shown in Figs. 8, 9, 10 and 11 indicate the fact that GSLB supports 46.2% of the traffic and IP Anycast support 5.9% of the traffic.

5.2 Stability of IP Anycast

In this study, we found 5.9% of data from/to Site_A is transferred via IP Anycast. However, given that IP Anycast requires stable routing, our expectation on IP Anycast was that it is mainly applicable for stateless service such as UDP. However, Fig. 17, which shows the average transferred byte for each IP Anycast transaction, clearly refutes such expectations and shows the high stability of the route. In other words, the size of the average transferred byte of each IP Anycast transaction indicates the stability of current internet route.

Figures 18, 19, 20 also show the stability of IP Anycast. Figure 18 shows the distribution of the number of packets by IP Anycast. As shown in Fig. 18, 80% of the IP Anycast transactions are two packets, i.e., one for request, and one for its response. However, 20% of transactions have three or more packets. Since some

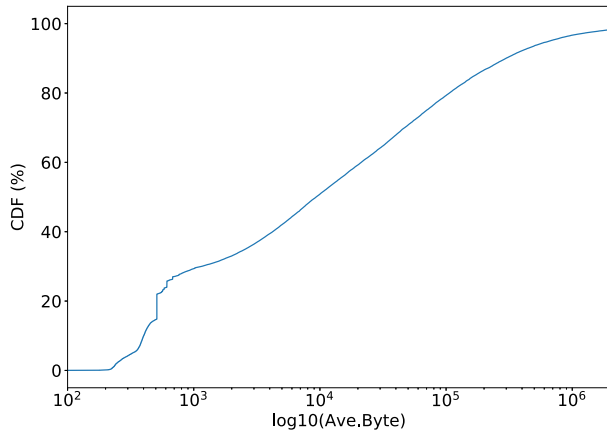


Fig. 19 Average transferred bytes for each destination.

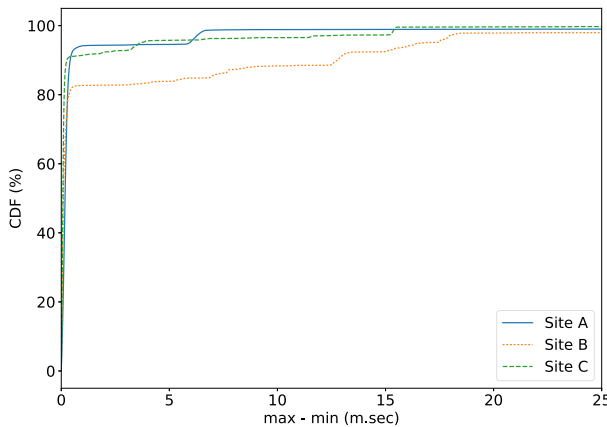


Fig. 20 Ping fluctuations (IP Anycast hosts).

transactions contain a large number of packets, the average transfer byte count for each destination IP is large. They are shown in Fig. 19. Here, while Fig. 17 shows the distribution with RTT information, Fig. 19 shows the CDF of packets from destinations with RTT less than 140 ms.

Figure 20 shows another fact of the IP Anycast's stability. While Fig. 15 shows the distribution of ping fluctuation, i.e., "max RTT - min RTT", of all destinations, Fig. 20 shows that of IP Anycast destinations. As clearly shown in Fig. 20, ping fluctuations of IP Anycast are small. 90% of fluctuations are less than 0.5 milliseconds for Site_A and Site_C. Although fluctuations for Site_B are large, still 80% of them are less than 0.4 milliseconds for Site_B. These small fluctuations show the stability of IP Anycast. Slightly large fluctuations for Site_B seems to be caused by the network condition inside North America.

5.3 Future Issues

Anderson et al. [21] reported that the latency is sometimes reduced when communicating through intermediate nodes due to BGP path selection. Our analysis ignores this issue. However, as shown in Fig. 8, use of different criterion makes only small differences. For example, using 100 ms (instead of 140 ms) as the criterion, 5.8% of byte-based traffic is classified as IP Anycast traffic. Note that ground distance between Site_A and Site_B is about 10,364 km. The speed of light in the optical cable is about 200,000 km/s ($\approx 299,792.458/1.5$, i.e., The speed of light in vacuum/Refractive index of quartz glass). Therefore, 100 ms is the

physical limit of RTT. Even if we use this physical limit, our analysis shown in Fig. 16 does not have significant errors. However, a detailed analysis of the related issues remains as one of the future research issues.

6. Conclusion

Recently, to achieve shorter response time, i.e., shorter RTTs, architectures such as Global Server Load Balancing (GSLB) and IP Anycast are being widely used. This study attempted to investigate how GSLB and IP Anycast contribute towards improving QoS from the users' perspective. The analysis of user traffic to and from our campus network led to the following findings:

- 93.8% of the traffic takes benefit from well operated internet services, i.e., 93.8% of the traffic has a Round-Trip Time (RTT) of less than 0.1 second.
- GSLB is still the more commonly used architecture for realizing fast response times. While GSLB supports approximately 46.2% of the traffic, IP Anycast supports 5.9% of the traffic.
- The length of the average transferred byte for each IP Anycast communication indicates the stability of the current internet route.

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