NAT64 vs SIIT: performance and scalability study for VoIP services

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ABSTRACT

The growing demand for IP addresses, driven by the proliferation of devices, has depleted the internet protocol (IP) version 6 (IPv6) reserves of some regional internet registries (RIRs). It is imperative to migrate to IPv6, offering an extended addressing space. This transition is no longer a choice but a necessity due to the exhaustion of IP version 4 (IPv4) addresses. The internet engineering task force (IETF) has implemented various transition strategies, such as the use of dual stack, IPv6-in-IPv4 tunnels, and address translation, due to the inconsistency between the two versions of the IP (IPv4 and IPv6). IPv4/IPv6 address translation mechanisms are crucial for the coexistence of networks using both protocols, with scalability playing a central role. Although these mechanisms offer advantages such as optimizing addressing space, their ability to scale effectively must be evaluated, especially in demanding scenarios such as voice over IP (VoIP). This article examines the scalability of two mechanisms, network address translation 64 (NAT64) and stateless IP/internet control message protocol (ICMP) translation (SIIT), in terms of VoIP clients in the graphical network simulator 3 (GNS3) environment. The results indicate that the SIIT mechanism is more performant and scalable than NAT64 in all measured parameters.

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1. INTRODUCTION

The communication between computer devices relies on a protocol that ensures the identification and recognition of the sources and destinations of data packets. Currently, internet protocol (IP) version 4 (IPv4), which uses a 32-bit address, is widely used, but its limited capacity of 4.3 billion devices becomes insufficient with the expansion of the internet. Faced with the scarcity of IP addresses among certain internet service providers (ISPs), the adoption of the recent version of the IP protocol has become significant. The internet engineering task force (IETF) designed IP version 6 (IPv6) with a focus on significant improvements in scalability, reliability, speed, and security, using a 128-bit address for its addressing space [1], [2].

Both of the versions IPv4 and IPv6 are fundamentally different, particularly in regards to IP address capacity and syntax. This disparity calls for a phased and incremental shift from IPv4 to IPv6. To ease this migration process, the IETF has introduced diverse mechanisms (dual stack, tunneling, and translation), each possessing unique characteristics [3].

The IPv4/IPv6 address translation mechanisms play an essential role in the migration from IPv4 to IPv6, enabling a harmonious coexistence of the two protocols. These mechanisms facilitate connectivity between networks using different IP versions, ensuring a smooth transition while preserving communication between IPv4 and IPv6 systems. They thus contribute to overcoming challenges related to IP address incompatibility, ensuring a gradual evolution to IPv6 while maintaining connectivity for existing infrastructures [4].

However, the issue of scalability of IPv4/IPv6 address translation mechanisms becomes concerning as IPv6 gains adoption, leading to an increase in the number of nodes and networks. The growing number of IPv6 clients and applications puts additional pressure on these translation mechanisms, potentially resulting in efficiency and administration. While research has been inclined to assess various translation mechanisms and identifying the best solution, a minority of research have examined the scalability's impact on their efficiency. A detailed study of the scalability of these mechanisms is essential to understand their response to increased IPv6 traffic, identify their scalability limits and constraints, and identifying methods to improve them to enable a seamless successful migration to IPv6, as the number of IPv6 customers and applications grows.

The paper describes an experiment carried out over graphical network simulator 3 (GNS3) to evaluate the performance and scalability of both translation techniques, network address translation 64 (NAT64) and stateless IP/internet control message protocol (ICMP) translation (SIIT). The research will take done over a GNS3-configured test network setup. Our strategy would entail raising the voice load by growing the number of clients using voice over IP (VoIP) communication, while also evaluating the both techniques to find the best answer.

The remainder of the paper is organized into the following sections: the second section describes the field research. Section 3 discusses our study's methodology and scenarios. Section 4 focuses on the evaluation technologies outcomes and analyses. The conclusion of this paper is presented in section 5.

2. RELATED WORKS

Due to the impending exhaustion of IPv4 addresses, which have become insufficient to meet the exponential expansion of connected devices, the migration to IPv6 poses a significant difficulty in computer networks. Because of its much wider address space, the adoption of the new protocol version, IPv6, is essential [5]. The examination of IPv6 routing is crucial for the success of the transition to IPv6. A detailed analysis of IPv6 routing is presented in [6], while [7] explores dynamic IPv6 routing protocols, detailing their principles, algorithms, and limitations. Evaluates the performance of intermediate system to intermediate system for IPv6 (IS-ISv6) routing in real-time applications such as voice and video, showing improved video throughput with average performance for voice [8]. Furthermore, Khadiri *et al.* [9] analyzes the influence of routing protocols (routing information protocol (RIPng), open shortest path first version 3 (OSPFv3), and IS-IS) on IPv4/IPv6 transition techniques, revealing that association of the IS-IS routing protocol provides significantly better performance in terms of latency and loss of packets, particularly in a video conferencing implementation.

Various studies have suggested different techniques to address the challenge of transitioning to IPv6, highlighting efficiency and performance as major concerns. Investigated numerous IPv6 transition mechanisms within a multiprotocol label switching (MPLS) network, determining that intra-site automatic tunnel addressing protocol (ISATAP) stood out for its excellent throughput and low jitter characteristics [10]. Jain and Payal [11], the 6 to 4 and manual tunneling strategies were configured with the RIPng and OSPFv3 routing protocols at the same time, revealing that 6 to 4 gives greater performance, particularly with the OSPF 6 to 4 network excelling in the majority of observed circumstances. Furthermore, the performance of three IPv4/IPv6 migration strategies (dual stack, manual, and 6 to 4 tunnel) have been examined in research [12], [13]. These assessments used VoIP and video conferencing. In terms of performance, the results showed that the dual-stack strategy outperformed tunneling technologies in terms of delay, mean opinion score (MOS) and jitter. D'yab and Lencse [14] created an experimental environment to investigate two IPv6 migration methods: lightweight 4 over 6 and dual stack lite. The usage of four virtual machines aided in the installation of each migration technique. The test setup was designed to perform benchmark assessments and compare the performance of two IP transition strategies per tunnel: dual stack lite as a stateful method and lightweight 4 over 6 as a stateless method.

According to the previously mentioned studies, many researchers have worked on security in IPv4/IPv6 transition scenarios. Concurrently, research on IPv6 routing and the performance measurement of specific transition approaches have been done. The scalability of the translation methods, however, has not been examined. Except for a single study on the scalability of tunneling mechanisms, namely manual tunneling and 6^{th} [15]. This study demonstrated that the 6^{th} tunnel is efficient and more scalable than the manual tunnel. Indeed, we emphasize the crucial importance of analyzing the scalability of these translation mechanisms because it defines the ability of a translation mechanism to withstand increased load, whether in terms of client count or IPv6 devices on the network. This paper gives a thorough examination of the performance and scalability of translation methods by expanding the number of customers participating in VoIP and experimenting the both methods to find the best. We will assess performance employing VoIP communication produced by IP service level agreement (SLA).

3. METHOD

We will investigate the performance and scalability of two IPv4/IPv6 address translation strategies, NAT64 and SIIT, in this part. We used the GNS3 to carry out this research application [16] to design a project enabling the creation and configuration of two separate scenarios for each technology: NAT64 and SIIT, as illustrated in Figures 1 and 2.

Figure 1. NAT64

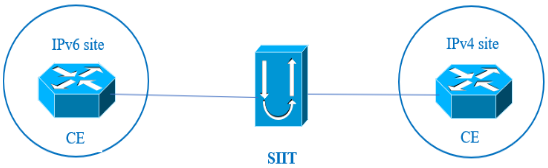

Figure 2. SIIT

3.1. Network address translation 64

It is a method that facilitates communication between devices using the IPv6 protocol and those using IPv4. Its operating principle is based on the conversion of addresses between IPv4 and IPv6, as shown in Figure 1. NAT64 is used when a device with an IPv6 address wishes to interact with a device with an IPv4 address, or vice versa. When an IPv6 packet reaches a network that is only compatible with IPv4, NAT64 steps in to translate the source IPv6 address into an IPv4 address. Similarly, when an IPv4 packet needs to be transmitted to a network compatible only with IPv6, NAT64 performs the translation by replacing the source IPv4 address with an IPv6 address [17], [18]. NAT64 operates as an address translator, enabling devices using different versions of the IP to communicate seamlessly. This method is critical in the migration from IPv4 to IPv6 by ensuring the smooth coexistence of both protocols, even when networks are heterogeneous in terms of supported IP versions [19].

3.2. Stateless IP/ICMP translation

The SIIT address translation technique is a method designed to enable communication between networks using IPv4 and IPv6, as illustrated in the Figure 2. Unlike NAT64, which employs stateful translation, SIIT is stateless, meaning it does not need to keep track of information about ongoing connections. The operation of SIIT is based on translating IP and ICMP headers between IPv4 and IPv6. When an IPv6 packet needs to be transmitted over an IPv4 network, SIIT replaces the IPv6 header with an IPv4 header while retaining relevant information. Similarly, when an IPv4 packet needs to be transmitted over an IPv6 network, SIIT performs the translation by replacing the IPv4 header with an IPv6 header [20].

The distinctive feature of SIIT is that it does not require state tracking for each connection, as translation is based on predefined rules. This simplifies implementation and improves efficiency but may limit some advanced features. SIIT thus contributes to the coexistence of both IP protocol versions within a network, facilitating the transition from IPv4 to IPv6 without requiring complex connection management [21]. There are other translation methods such as transport relay translator (TRT) [22], Socks-Based IPv6/IPv4 Gateway (SOCKS64) [23], and bump in the stack (BIS) [24].

Opting for VoIP traffic as the background traffic, we chose to use IP SLA [25] to create this traffic and evaluate the connection's and applications quality. IP SLA, a Cisco approach, enables the production of traffic to test across different network devices. The goal of this research is to create communication between

IPv6 and IPv4 sites employing both translation methods (NAT64 and SIIT), while also testing their scalability by increasing the traffic, as measured by the number of customers communicating over VoIP.

With this project, we created 104 unique scenarios. For each technology (NAT64 and SIIT), we progressively raised the number of consumers who engaged in VoIP communications, from two to 100. Tables 1-3 show the setup and performance assessment details. Table 1 represents for each scenario, the routing and addressing are employed. Table 2 illustrates the parameters of the VoIP traffic generated by IP SLA used for the performance and scalability evaluation of two mechanisms: NAT64 and SIIT. Table 3 describes the performance measurement parameters for the two mechanisms NAT64 and SIIT.

Table 1. For each scenario, routing, and addressing are employed

Scenarios	Addressing		Routing	
	NAT64	SIIT	NAT64	SIIT
IPv4 site	Addressing IPv4		RIPv2	
IPv6 site	Addressing IPv6		RIPng	

Table 2. Parameters of the VoIP traffic generated by IP SLA

Used traffic	VoIP
Used codec	G729
Count of packets	1000 packets
Time separating packets	20 milliseconds

Table 3. Performance measurement parameters

Delay	Represents the time it takes a packet to travel from where it started to its receiving at the destination
Jitter	Corresponds to the variation in end-to-end delay between packets inside the same data stream. The optimal scenario aims for minimal jitter, approaching zero.
MOS score	Plays a vital role in assessing voice application quality, the MOS changes according on the codec used, and is rated from 0 to 5, with 5 signifying good quality and 1 indicating low quality. For example, under perfect conditions, the MOS for the G.729 codec can reach 4.06.
Packet loss rate	Measured as a percentage, reflects the proportion of lost packets in comparison to the total sent. This crucial metric gauges network or communication reliability by indicating the percentage of packets that failed to reach their intended destination.

4. RESULTS AND DISCUSSION
4.1. Analysis of the findings
4.1.1. Delay

The findings shown in Figure 3 offer delay measurements in milliseconds, for each evaluated technology, namely NAT64 and SIIT. These latency values are crucial for assessing the performance and efficiency of these communication mechanisms. The results show that as the number of customers communicating via VoIP increases, the SIIT mechanism remains superior to NAT64. This performance disparity is explained by the distinctive strategy of each mechanism. Firstly, SIIT operates in a stateless manner, meaning it does not retain information about connections. This can lead to a simplification of the address translation process, potentially reducing delays associated with connection management. Additionally, SIIT does not need to maintain a session state, unlike NAT64, which can operate in stateful mode. The absence of state management can contribute to a faster response in translating packets between IPv6 and IPv4 networks.

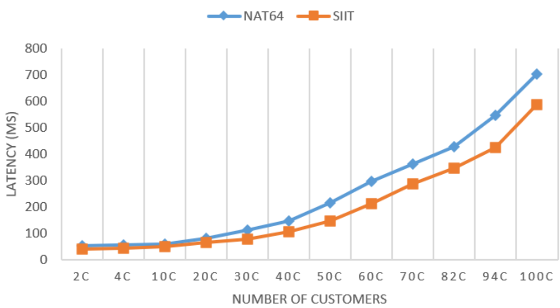

Figure 3. Delay

4.1.2. Jitter

The jitter results are presented in Figure 4. This criterion corresponds to the variation in end-to-end delay between packets. These results show that the SIIT address translation mechanism outperforms NAT64, displaying lower jitter values than the latter.

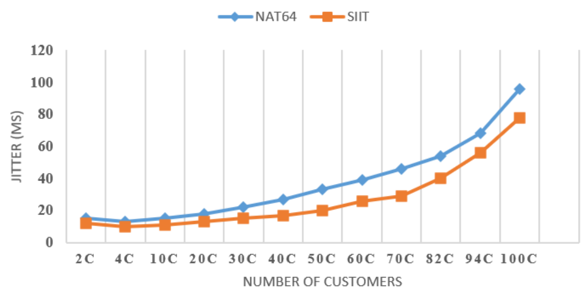

Figure 4. Jitter

4.1.3. Packet loss rate

The results for the rate of packet loss are shown in Figure 5. When the VoIP load is raised, the results show that the SIIT method has a lower packet loss rate than NAT64. This is explained by SIIT's improved packet management as load grows, resulting in a decreased packet loss rate when compared to the NAT64 approach.

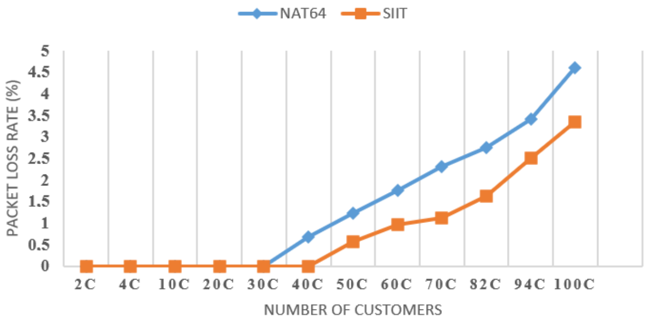

Figure 5. Packet loss rate

4.1.4. Mean opinion score

The MOS linked with the researched methods are depicted in Figure 6. A higher MOS indicates greater voice quality performance. The conclusions drawn from this figure indicate that the SIIT address translation mechanism provides superior voice quality compared to NAT64.

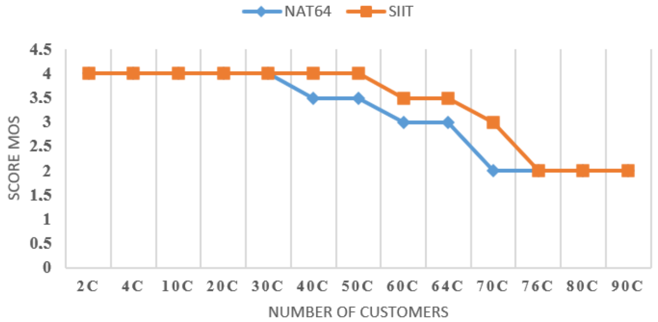

Figure 6. MOS

4.2. Scalability debate

We evaluated the scalability of two address translation systems, SIIT, and NAT64, in this study utilizing the GNS3 tool. Each of those methods was tested in 52 different situations, each of which involved a rise in the number of customers engaging in VoIP communications, ranging from two to 100 customers. Our findings show that when the number of VoIP customers increases, the SIIT address translation method beats the NAT64 mechanism for all examined measurement metrics. This superiority is explained by the stateless nature of SIIT, which does not retain information about connections. This characteristic simplifies the address translation process, potentially reducing delays associated with connection management. Thus, SIIT, operating without maintaining a session state, provides enhanced performance compared to NAT64 in terms of latency, jitter, packet loss, and MOS, contributing to quicker responses in packet translation between IPv6 and IPv4 networks.

To evaluate the scalability of the investigated methods, it is now necessary to include acceptable delay (400 ms), jitter (50 ms), and rate of loss (3%) levels at the same time. This will assist identify whether method can survive the increased load, as measured by the number of customers engaging via VoIP. While the NAT64 method has a reasonable loss rate of 2.76% in the scenario with 82 customers, its delay in the same context reaches 426 ms, making VoIP unusable from this scenario. Up to 94 clients, the SIIT mechanism maintains a tolerable loss rate of 2.52%. In this same case, however, the delay reaches 423 ms, rendering VoIP ineffective from that point forward.

Indeed, Figure 7 presents a detailed evaluation of the scalability of the analyzed methods, notably the NAT64 and SIIT processes, in light of the growing number of VoIP customers. This comprehensive evaluation enables us to better understand the behavior of these technologies under growing load, evaluating their capacity to maintain acceptable performance while taking these crucial service quality elements into account. Based on these findings, it is clear that the SIIT method outperforms NAT64 in terms of scalability.

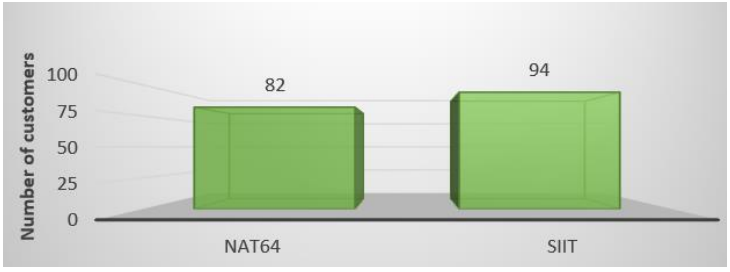

Figure 7. Scalability

5. CONCLUSION

We tested and assessed the performance and scalability of address translation methods, namely NAT64 and SIIT, in this research by increasing the VoIP load in terms of the number of customers participating. This GNS3 study relies on four measurement criteria: dealy, jitter, MOS, and loss of packets. The results show that the SIIT method has better scalability than NAT64, outperforming it in all measurement metrics. The main objective was to assess which translation technique provides better performance and scalability as network load increases in terms of the number of VoIP customers. Future perspectives could focus on studying the security of these translation mechanisms to identify potential vulnerabilities and propose solutions. This work has the potential to make significant contributions to network administrators, policymakers, and researchers seeking to enhance the security of their networks and prevent attacks.

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