VoIP Over WAN Performance Testing

A CAPSTONE PROJECT REPORT

Submitted in the partial fulfilment for the Course of

CSA0735 – Computer networks for communication

to the award of the degree of

BACHELOR OF ENGINEERING

IN

AIDS, CSE, AIDS

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DECLARATION

We**, LATISHA S, MOUNNILA S P, SAI LIKHITH K** of the **AIDS, AIDS**, **CSE (CORE)** Saveetha Institute of Medical and Technical Sciences, Saveetha University, Chennai, hereby declare that the Capstone Project Work entitled **VoIP Over Wan Performance Testing** is the result of our own bonafide efforts. To the best of our knowledge, the work presented herein is original, accurate, and has been carried out in accordance with principles of engineering ethics.

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BONAFIDE CERTIFICATE

This is to certify that the Capstone Project entitled **"VoIP OVER WAN PERFORMANCE TESTING"** has been carried out by **LATISHA S, MOUNNILA S P, SAI LIKHITH K** under the supervision of **Dr RAJARAM P** and is submitted in partial fulfilment of the requirements for the current semester of the B.Tech **AIDS** program at Saveetha Institute of Medical and Technical Sciences, Chennai.

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ABSTRACT

Voice over IP (VoIP) over Wide Area Networks (WAN) presents unique challenges due to factors such as latency, jitter, packet loss, and bandwidth limitations, which can degrade call quality and reliability. Performance testing is crucial to evaluate and optimize VoIP service quality before deployment in enterprise or service provider environments. This paper examines the key methodologies and metrics for assessing VoIP performance over WAN, including latency, jitter, packet loss, Mean Opinion Score (MOS), and bandwidth utilization.

The study explores various testing approaches, such as simulated traffic generation, real-world scenario emulation, and Quality of Service (QoS) policy validation, using industry-standard tools like Wireshark, Iperf, and specialized VoIP analyzers. Additionally, the impact of different codecs (e.g., G.711, G.729) and network conditions on voice quality is analyzed.

The findings highlight best practices for mitigating performance issues, ensuring optimal VoIP service delivery over WAN. This research provides a framework for network engineers and IT professionals to conduct effective VoIP performance testing, enhancing user experience and operational efficiency.

Keywords: VoIP, WAN, Performance Testing, QoS, Latency, Jitter, Packet Loss, MOS, Bandwidth Optimization.

CHAPTER-1

INTODUCTION

Voice over IP (VoIP) has become a critical communication technology for businesses and service providers, enabling cost-effective and flexible voice communications over IP networks. However, when VoIP is deployed over a Wide Area Network (WAN), performance challenges such as latency, jitter, packet loss, and bandwidth constraints can significantly impact call quality and reliability.

Fig.1.1

Performance testing for VoIP over WAN is essential to ensure optimal voice quality, assess network readiness, and identify potential bottlenecks before deployment. This testing involves simulating real-world traffic conditions to measure key Quality of Service (QoS) metrics, including:

- **Latency (Delay):** The time taken for voice packets to travel from sender to receiver. Excessive delay can cause conversational difficulties.
- **Jitter:** Variations in packet arrival times, leading to uneven voice playback.
- **Packet Loss:** Missing voice packets due to network congestion or errors, resulting in degraded audio quality.
- **Mean Opinion Score (MOS):** SA standardized metric (1-5 scale) that quantifies perceived voice quality.
- **Bandwidth Utilization:** Ensuring sufficient bandwidth to support VoIP traffic alongside other data.

Fig.1.2

By conducting rigorous performance testing, organizations can optimize WAN configurations, implement QoS policies, and select appropriate codecs to deliver a seamless VoIP experience. This introduction explores the methodologies, tools, and best practices for evaluating VoIP performance over WAN environments.

PERFORMANCE OF VOIP OVER WIDE AREA NETWORK

Fig.1.3

- Jitter Impact
  Fluctuating packet arrival times (jitter) can cause choppy or distorted audio.
- Latency Sensitivity
  VoIP requires low latency (≤150 ms one-way); high latency disrupts conversation flow.
- Packet loss
  Even minimal packet loss (≥1%) leads to voice breakups and poor call quality.
- Bandwidth Availability
  VoIP performance degrades if there isn't sufficient or consistent bandwidth.
- Codec Selection
  Codecs like G.711 (high quality, high bandwidth) and G.729 (compressed, low bandwidth) affect call clarity and data usage.
- Quality of Service (QoS)
  QoS settings prioritize VoIP traffic over data, ensuring timely voice packet delivery.
- Call Setup Delay
  Slow SIP signalling over WAN can increase the time taken to establish a call.
- Network Congestion
  High traffic on WAN can cause jitter, delay, and packet loss, degrading VoIP performance.
- Firewall
  VoIP may struggle over firewalled or NAT-ed WANs unless protocols like STUN, TURN, or ICE are used.
- Route Variability
  Multiple hops and changing routes in WAN can introduce delay and packet reordering.
- Echo and Feedback
  Long WAN delays can cause echo, requiring echo cancellation measures.

- Protocol Overhead
  SIP and RTP protocols introduce overheads that slightly reduce performance.
- Call Drop Risk
  Unstable WAN links can cause mid-call disconnections or dropped calls.
- Security Effects
  Encryption (e.g., SRTP) can increase processing delay slightly but is essential for secure calls.
- Redundancy and Failover
  Lack of backup routes or VRRP-like redundancy may lead to communication outages.
- Real-Time Monitoring Needs
  Tools like Wireshark or VoIP monitor are essential to diagnose live performance issues.
- Codec Mismatch Across Devices
  If endpoints use incompatible codecs, transcoding is needed, reducing performance.
- Concurrent Call Load
  Multiple simultaneous VoIP calls over WAN require more bandwidth and management.
- Call Quality Metrics
  MOS (Mean Opinion Score) and R-Factor scores help measure real user experience.

Fig.1.4

CHAPTER-2

EVALUATION OF VoIP OVER WAN

- **Cost Efficiency**
  VoIP over WAN reduces telecommunication costs by using existing data networks instead of traditional phone lines.
- **Flexibility and Scalability**
  Easily supports remote offices and mobile users across geographical locations without needing separate infrastructure.
- **Voice Quality Depends on Network Conditions**
  Call clarity is highly dependent on latency, jitter, packet loss, and bandwidth across the WAN.
- **Latency Issues**
  Longer geographic distances and more network hops in WAN can introduce significant voice delay.
- **Jitter and Voice Distortion**
  Variable packet arrival times in WAN links often cause jitter, affecting audio smoothness unless mitigated by jitter buffers.
- **Packet Loss Challenges**
  Poor WAN links may drop packets, resulting in voice clipping, silence, or robotic sound quality.
- **Quality of Service (QoS) Requirement**
  Proper QoS must be configured to prioritize voice traffic, or data traffic may interfere with VoIP performance.
- **Codec Efficiency**
  Compressed codecs (like G.729) help reduce WAN bandwidth usage but may slightly lower audio quality compared to G.711.
- **Call Setup Delays**
  SIP signal over WAN may experience delays due to latency or firewall traversal.
- **NAT and Firewall Traversal**
  WAN routers using NAT may block SIP or RTP traffic unless STUN, TURN, or ICE is implemented correctly.
- **Security Overheads**
  Securing VoIP over WAN with SRTP or VPN adds encryption overhead, which can slightly affect performance.
- **Reliability and Failover Support**
  Without redundancy (e.g., using VRRP or secondary WAN links), VoIP calls may drop during network failure.

- **Monitoring Complexity**
  Troubleshooting VoIP over WAN requires detailed monitoring tools like Wireshark, SolarWinds, or VoIP monitor.
- **Network Congestion Impact**
  Shared WAN links may suffer during peak data usage, leading to voice quality degradation.
- **End-to-End Delay**
  Calls across continents may experience noticeable delays due to increased hop count and propagation time.
- **Performance Varies Across Locations**
  VoIP quality may differ between sites depending on the local WAN link quality and ISP performance.
- **Call Consistency**
  VoIP over a stable WAN offers near landline quality, but unstable links result in call drops and jitter.
- **Integration with PBX Systems**
  VoIP can be seamlessly integrated with cloud or on-premise PBX systems, improving unified communication.
- **Mobile VoIP Challenges**
  WAN links through mobile networks (e.g., 4G/5G) may fluctuate, impacting voice quality unpredictably.
- **Overall Effectiveness**
  With proper configuration, monitoring, and quality WAN links, VoIP over WAN is an effective, scalable, and modern communication solution.

Fig.2.1

ANALYSIS OF VOIP CALL QUALITY ACROSS WAN USING ASTERISK AND SIP

- **Bandwidth Optimization**
  Analysis includes how bandwidth limitations in WAN affect multiple concurrent VoIP streams.
- **Echo & Audio Quality**
  Testing involves detecting echo and evaluating Asterisk's echo cancellation effectiveness.
- **Security Implications**
  Use of SIP over TLS and SRTP is tested for securing signaling and media without degrading performance.
- **Call Load Testing**
  Multiple concurrent calls are placed to assess scalability and performance degradation under WAN stress.
- **Overall Result**
  With correct configuration and QoS, VoIP over WAN using Asterisk and SIP can deliver reliable and high-quality communication.

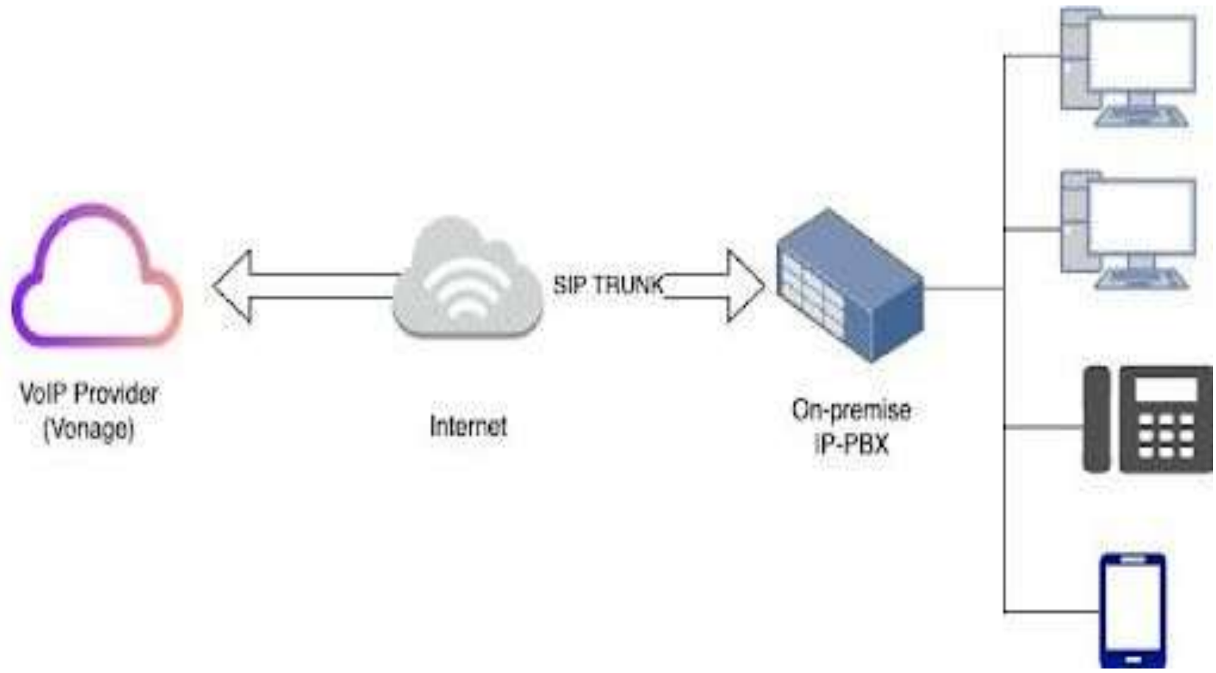

Fig.2.2

CHAPTER-3

TESTING OF VOIP PERFORMANCE IN WAN SETUP

- **Bandwidth Testing**
  Verify if available WAN bandwidth supports the expected number of concurrent VoIP calls.
- **Latency Measurement**
  Use tools (like ping, traceroute) to measure one-way and round-trip delay; acceptable latency is $\leq$150 ms.
- **Jitter Analysis**
  Test jitter using VoIP test tools; values above 30 ms may cause audio distortion.
- **Packet Loss Detection**
  Simulate calls and monitor for packet loss; VoIP requires <1% packet loss for acceptable quality.
- **Call Quality Assessment (MOS)**
  Use tools like VoIP monitor or Wireshark to determine MOS (Mean Opinion Score); ideal is $\geq$4.0.
- **Codec Efficiency Testing**
  Evaluate different codecs (G.711, G.729, Opus) under WAN constraints for quality vs. bandwidth usage.
- **SIP Registration and Signal Test**
  Test SIP message exchange (INVITE, ACK, BYE) for successful call setup and teardown.
- **Firewall and NAT Traversal Testing**
  Ensure SIP and RTP traffic passes through NAT and firewall using STUN, TURN, or ICE protocols.
- **Call Setup Time Testing**
  Measure time between dial and connection; should ideally be <3 seconds in a healthy WAN.
- **QoS Configuration Verification**
  Ensure routers/switches prioritize VoIP traffic using DSCP or VLAN tagging.
- **RTP Stream Analysis**
  Capture RTP streams and evaluate timing, jitter, sequence number issues, and delays.
- **Concurrent Call Load Test**
  Simulate multiple VoIP calls to measure performance and stability under load.
- **Failover and Redundancy Test**
  Test behavior during WAN link failures or switchovers using backup routes or VRRP.
- **Echo and Audio Delay Testing**
  Verify audio clarity and absence of echo using loopback test calls.

- **Voice Delay Across Hops**
  Measure delay introduced by each intermediate router/switch in the WAN path.
- **Real-Time Monitoring Setup**
  Use tools like PRTG, Nagios, or Zabbix to continuously monitor VoIP performance metrics.
- **Security and Encryption Impact**
  Test performance with SRTP and SIP over TLS to ensure security doesn't impact call quality.
- **Call Drop Analysis**
  Monitor for unexpected call terminations and identify root causes (e.g., timeout, link drops).
- **Softphone/IP Phone Compatibility Test**
  Test different endpoints for codec support, SIP registration, and call handling.
- **Detailed Log and Report Generation**
  Record all test results including call logs, packet captures, and MOS trends for analysis and documentation.

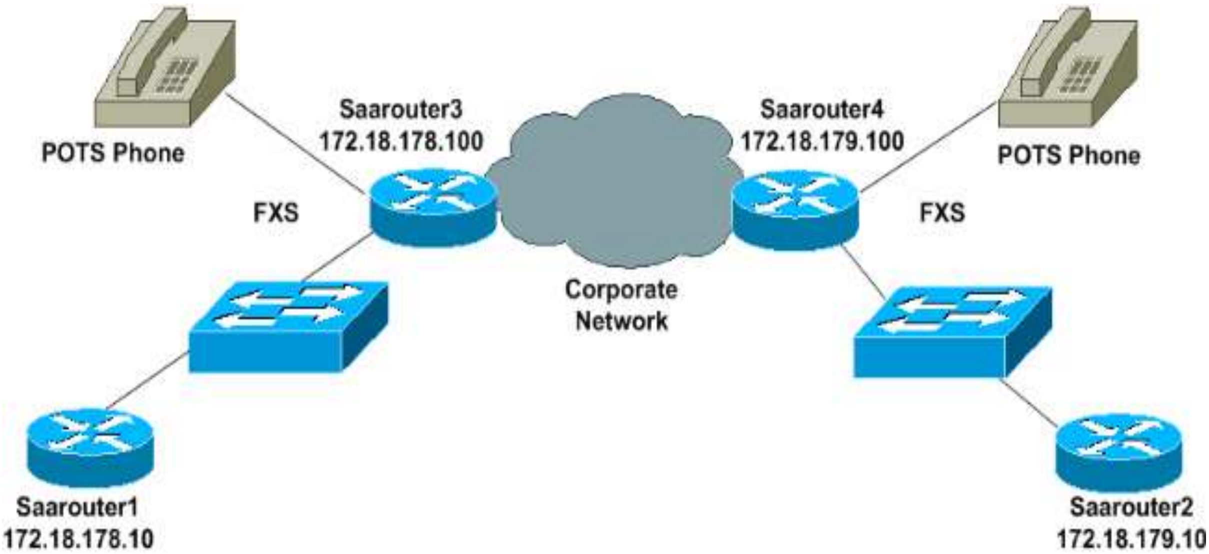

Fig.3.1

PROBLEM OCCURRED DURING VOIP OVER WAN

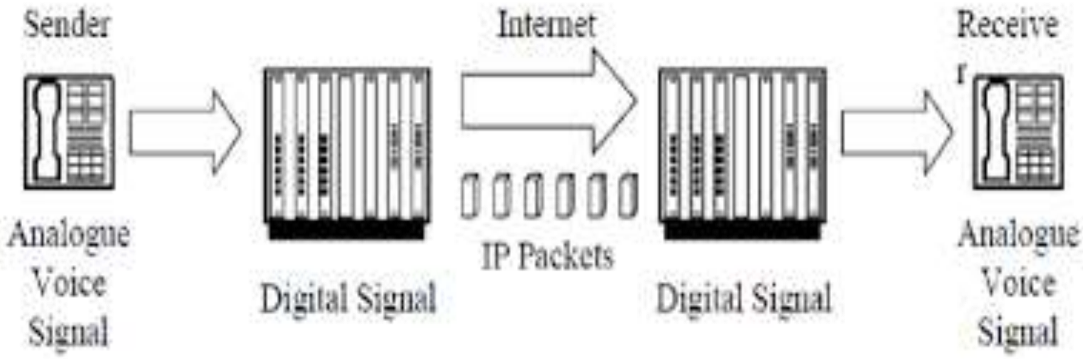

Fig.3.2

High Latency
Delays in voice transmission make conversations feel unnatural and out of sync.

Jitter (Delay Variation)
Irregular packet arrival times result in choppy or scrambled audio during calls.

Packet Loss
Missing voice packets lead to broken speech, silence, or robotic voice effects.

Poor Bandwidth Availability
Insufficient bandwidth causes call quality degradation, especially during peak network usage.

Call Dropping
Unstable WAN links or routing failures can cause active calls to drop unexpectedly.

Codec Mismatch or Overload
Using high-bandwidth codecs on a low-speed WAN leads to audio clipping or dropped calls.

SIP Registration Failures
Firewalls or NAT configurations may block SIP messages, preventing successful call setup.

NAT and Firewall Traversal Issues
Improper configuration can block RTP or SIP traffic, causing one-way audio or failed calls.

SOLUTION

To overcome the common problems experienced during VoIP communication over WAN, several network-level and configuration-based solutions must be implemented. High latency issues can be minimized by using dedicated WAN links, optimizing routing paths, and avoiding long detours in packet delivery. To address jitter, implementing jitter buffers on VoIP endpoints helps absorb delay variations and smooth audio playback. Packet loss can be reduced through QoS (Quality of Service) configurations that prioritize VoIP packets over regular data traffic, ensuring reliable delivery even during network congestion. Insufficient bandwidth can be resolved by upgrading WAN capacity or compressing voice data using low-bandwidth codecs like G.729.

For call dropping problems, ensuring link stability, using redundant paths with VRRP or BGP, and keeping SIP session timers optimized can provide reliability. Codec mismatches should be prevented by standardizing codecs across devices and configuring transcoding only where necessary. SIP registration failures and call setup issues caused by NAT and firewalls can be fixed using STUN, TURN, or ICE protocols that enable secure NAT traversal. Firewalls should be configured to allow SIP (port 5060) and RTP (typically ports 10000–20000) traffic. QoS must be properly enabled on routers and switches to mark and prioritize VoIP packets using DSCP or VLAN tagging. Lastly, to eliminate echo and voice delays, echo cancellation settings should be enabled on devices, and the overall delay should be reduced by streamlining the WAN routing path. With these measures in place, VoIP over WAN can perform efficiently and deliver high-quality voice communication.

CONCLUSION

VoIP communication over WAN often encounters several issues, but these can be effectively resolved with proper network configuration and system optimization. One of the primary problems is high latency, which can be reduced by optimizing routing paths and using faster, dedicated WAN connections. Jitter, which causes choppy audio, can be managed by implementing jitter buffers at the receiving endpoints. Packet loss, another major concern, is minimized by applying Quality of Service (QoS) settings that prioritize voice packets over less critical data. In cases where bandwidth is insufficient, upgrading WAN capacity or using compressed codecs like G.729 can help maintain call quality. Call drops due to unstable links can be prevented through redundancy mechanisms such as VRRP and careful SIP timer configuration. NAT and firewall issues that block SIP or RTP traffic can be solved using protocols like STUN, TURN, or ICE, along with proper port forwarding settings.

To prevent codec mismatch, all devices should support the same set of codecs, and transcoding should be limited to essential cases. QoS must be consistently configured across all routers and switches in the WAN to ensure voice traffic is given the highest priority. Lastly, echo and voice delay issues can be resolved by enabling echo cancellation features and reducing network delay through improved routing. By addressing these problems proactively, VoIP over WAN can achieve reliable, clear, and efficient communication.

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