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Study of Video Conferencing Techniques and Performance Evaluation of Skype

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Abstract

The evolution of digital communication has significantly transformed enterprise collaboration, with EO (Enterprise Online) conferencing emerging as a cornerstone for virtual meetings, remote learning, and global teamwork. Among the various tools available, Skype has been a pioneer in providing voice, video, and instant messaging services across diverse platforms. This report presents a comprehensive investigation into EO conferencing techniques, with a particular focus on the architectural design, operational mechanisms, and performance evaluation of Skype in enterprise environments. The study employs real-time testing and simulation-based methods to assess Skype's performance across critical parameters including audio and video quality, latency, jitter, packet loss, throughput, connection reliability, and bandwidth utilization. Comparative benchmarking against other popular platforms such as Zoom and Microsoft Teams highlights Skype's relative performance strengths and constraints. The findings reveal that while Skype offers reliable functionality and ease of use, its performance may degrade under high-load or low-bandwidth conditions. This report provides actionable insights for IT managers and organizations seeking efficient EO conferencing solutions and outlines future research directions to enhance scalability, security, and user experience in digital conferencing ecosystems.

INTRODUCTION

In the era of globalization and digital transformation, Enterprise Online (EO) conferencing platforms have become essential for real-time interaction across geographically dispersed teams. Skype, a prominent EO conferencing platform, has evolved from a peer-to-peer voice calling tool to a multi-functional communication tool that supports voice and video conferencing, screen sharing, file transfer, and instant messaging. Despite facing competition from newer entrants like Zoom, Microsoft Teams, and Google Meet, Skype continues to be used by millions due to its simplicity, stability, and wide compatibility.

The performance of EO conferencing tools has come under increased scrutiny due to rising expectations for high-definition video, low-latency audio, and uninterrupted connectivity. Factors such as network congestion, bandwidth limitations, device hardware, and application architecture all play crucial roles in determining the quality of user experience. This report aims to explore the underlying techniques of EO conferencing and conduct a comprehensive performance evaluation of Skype, examining key metrics such as latency, jitter, packet loss, audio/video clarity, connection stability, and bandwidth consumption. Comparative benchmarking against other leading EO conferencing platforms is also conducted to contextualize Skype's capabilities.

In the modern era of globalization and digital transformation, video conferencing has become a crucial innovation, playing a crucial role in various sectors, including business, education, healthcare, governance, and personal communication. Skype, developed by Microsoft, offers voice calls, video calls, instant messaging, screen sharing, and file transfers over a peer-to-peer architecture. This study aims to explore the core techniques used in video conferencing systems, analyze the architecture and underlying technologies of Skype, and conduct a performance evaluation based on key parameters.

Figure 1: Image of Live Meet

Video conferencing is the transmission of audio and video signals in real time, allowing multiple users to interact visually and audibly. It relies on various core technologies such as audio/video compression (codecs), transport protocols, buffering mechanisms, and real-time communication (RTC) frameworks. There are two primary architectures for video conferencing: peer-to-peer (P2P) and client-server (MCU based). Key technologies in video conferencing include codecs like H.264, VP8/9, and Opus, protocols like RTP/RTCP, SIP, and WebRTC, and Quality of Service (QoS) mechanisms to manage latency, jitter, and packet loss.

Skype, developed by Microsoft, initially operated using a proprietary P2P protocol called Skype Protocol, which evolved over time into a hybrid architecture integrating Microsoft's cloud services. Key features include voice and video communication using dynamic codecs, NAT Traversal Techniques to connect users behind firewalls or NATs, encryption and security using AES encryption and secure signalling mechanisms, and super node architecture. In recent versions, Skype shifted toward a more centralized architecture post-Microsoft acquisition, improving reliability, especially for group calls and integration with cloud services like Microsoft Teams and Office 365.

The performance of a video conferencing tool like Skype is evaluated using several criteria: delay, jitter, bandwidth usage, video/audio quality, reliability and uptime, and user experience. The evolution of communication technologies has made it possible to transmit not just voice but also high-quality video across networks, making video conferencing systems a cornerstone of global communication.

A successful system must seamlessly integrate software components, hardware components, and network infrastructure. Core components and technologies involved include codecs (coders/decoders), communication protocols (RTP, RTCP, SIP, WebRTC), transport layer support, and Network Address Translation (NAT) Traversal.

Skype emerged as a disruptive technology in internet-based communication in 2003, introducing a Peer-to-Peer (P2P) architecture that allowed users to directly connect without relying on centralized servers. It became popular for free voice and video calls over the internet, instant messaging and file sharing, group video conferencing and screen sharing, and cross-platform compatibility.

Evaluating Skype's performance under different scenarios is vital to understand how well it meets modern communication needs, identify its strengths and weaknesses, and recommend improvements or alternative platforms. This background provides a foundation for exploring video conferencing technologies in greater depth, analyzing Skype's operational model, and systematically evaluating its performance.

Problem Statement

For video conferencing a large area of packet data is exchanged between users. There is unreliable transmission of packets, for these we have to know about the video conferencing parameters. The quality of packet, how fast the packet can travel, also the quality of video is necessarily be known to improve the performance. This can be done by studying video conferencing technique and parameter evaluation of Skype.

Summary of Reviews

With the surge in remote communication due to the COVID-19 pandemic and beyond, video conferencing applications have become a critical

means for business, education, and healthcare delivery. Various researchers have explored the performance analysis and optimization of video conferencing systems across network and application layers.
S. Singh and R. Bansal (2020) emphasized the importance of Quality of Service (QoS) metrics such as jitter, packet loss, and latency in real-time applications like Zoom and Google Meet, demonstrating that these parameters are directly influenced by underlying network protocols like UDP and TCP [Singh & Bansal, 2020].
K. V. Mahalakshmi et al. (2021) carried out a comparative study on video conferencing platforms using Wireshark analysis, focusing on packet capture (PCAP) data. They highlighted the use of UDP in minimizing delay, and TCP's role in ensuring data reliability during file transfers within conferencing tools [Mahalakshmi et al., 2021].
J. Kim and M. Kim (2020) addressed the adaptive bitrate streaming techniques used by platforms like Microsoft Teams and Skype. Their work showed how dynamic encoding adapts to fluctuating bandwidth, improving user experience without overloading the network [Kim & Kim, 2020].
R. Sharma and D. Kumar (2022) analyzed real-time packet behavior in conferencing applications, revealing how voice activity detection (VAD) and packet size variation are strong indicators of system health. Their packet loss and retransmission studies also suggested thresholds beyond which video quality degrades [Sharma & Kumar, 2022].
P. Patil and S. Joshi (2019) focused on security aspects, showing how IP filtering and protocol identification help detect anomalies like spoofing or DoS attacks in conferencing sessions, stressing the importance of forensic packet analysis [Patil & Joshi, 2019].
Furthermore, S. Ahmad et al. (2021) discussed the integration of deep packet inspection (DPI) and AI-based tools in analyzing conferencing apps. They proposed enhancements in traffic classification and prioritization, especially for hybrid work environments relying on Zoom or Cisco WebEx [Ahmad et al., 2021].
Farreras et al. (2025) introduce GNNetSlice, a Graph Neural Network–based model designed to predict network slice performance in 5G and Beyond-5G (B5G) core and transport networks. By modeling slice-specific traffic flows as graph structures, their data-driven approach achieves highly accurate performance predictions—SMAPE of 5.22 % for delay, 1.95 % for jitter, and 2.04 % for packet loss—while maintaining near-real-time inference suitable for dynamic resource allocation decisions ResearchGate. This work fills a critical gap in GNN-based KPI prediction for network slicing and offers a publicly available model and dataset on GitHub for further experimentation.
Guo et al. (2025) propose the Adaptive Cross-Layer Optimization Transmission Method with Environment Awareness (ACOTM-EA) to improve streaming performance in high-speed mobile scenarios—specifically high-speed rail networks. The method combines environmental sensing with predictive modeling of channel quality and proactive base station handoffs. Evaluated in realistic conditions, ACOTM-EA demonstrated an 11 % throughput improvement and a 5 % boost in perceived media quality for passengers. Their future work highlights AI-driven cross-layer scheduling and hybrid network collaboration techniques.

Summary Table

Study / Authors	**Focus Area**	**Key Contributions**
Farreras et al. (2025)	GNN-based network slice performance model	High accuracy KPI prediction, scalable for B5G slices
Guo et al. (2025)	Environment-aware streaming optimization	Throughput & QoE improvements in high-speed mobility
Smirnov & Tomforde (2024)	DRL-based WebRTC rate control in 5G	Better QoE vs standard congestion control

Smirnov & Tomforde (2024) present a model-free Deep Reinforcement Learning (DRL) approach to enhance uplink rate control in WebRTC-based video streaming over 5G. Their DRL agent dynamically adjusts sending bitrate using WebRTC-compliant metrics and outperformed the default Google Congestion Control (GCC) algorithm in simulations—delivering better throughput while maintaining comparable delay and packet loss levels. The architecture is explicitly designed for real-world integration and future extension with hybrid learning policies combining simulated and real testbed data.
These three studies collectively illustrate emerging trends in network-aware optimization, real-time performance modeling, and reinforcement learning–based rate control in next-generation multimedia and slicing architectures. Let me know if you'd like deeper integration into your literature review, including

comparative analysis frameworks or methodology evaluation.

Research Gap Analysis

Despite numerous studies on EO conferencing systems and Skype's performance, several gaps remain. Firstly, most existing evaluations of Skype are outdated, conducted before significant architectural transitions to cloud infrastructure. Secondly, few studies provide a head-to-head comparison of Skype with newer tools under controlled network scenarios. Thirdly, while technical metrics are often measured, user experience—especially in enterprise-scale deployments—is underexplored. Additionally, there is limited research on Skype's behavior under hybrid network conditions (e.g., fluctuating bandwidth, mobile hotspot environments). These gaps highlight the need for an updated, holistic study that combines technical benchmarking with practical usability analysis to assess Skype's viability as a modern EO conferencing solution.

- Lack of comparative performance analysis between Skype and newer video conferencing platforms like Zoom, Google Meet, and Microsoft Teams.
- Limited studies focusing on Skype's performance under different network conditions (e.g., low bandwidth or high latency environments).
- Inadequate research on the security and privacy aspects of Skype's communication protocols.
- Insufficient real-time user experience evaluation based on modern quality metrics (e.g., MOS – Mean Opinion Score).
- Few studies have addressed the integration of Skype in hybrid work or educational settings post-COVID-19.
- Minimal focus on energy consumption and resource usage during video conferencing sessions.
- Lack of updates on Skype's technical performance after Microsoft's platform integration and updates.
- Underexplored area of how Skype performs on mobile vs. desktop environments in terms of video quality and stability.

Goal of Work

The aim of this study is to investigate and evaluate the techniques used in Enterprise Online (EO) conferencing, with a specific focus on assessing the technical performance, operational efficiency, and user experience of the Skype platform under various network and usage conditions.

RESEARCH METHODOLOGY

This research adopts a quantitative, experimental design focused on analyzing the performance of Skype as an EO conferencing platform under controlled and real-world conditions. The goal is to measure and evaluate Skype's technical behavior based on various network variables and user load levels.

Data is collected systematically using performance monitoring tools and analyzed using statistical methods to interpret Skype's capability and consistency across communication metrics.

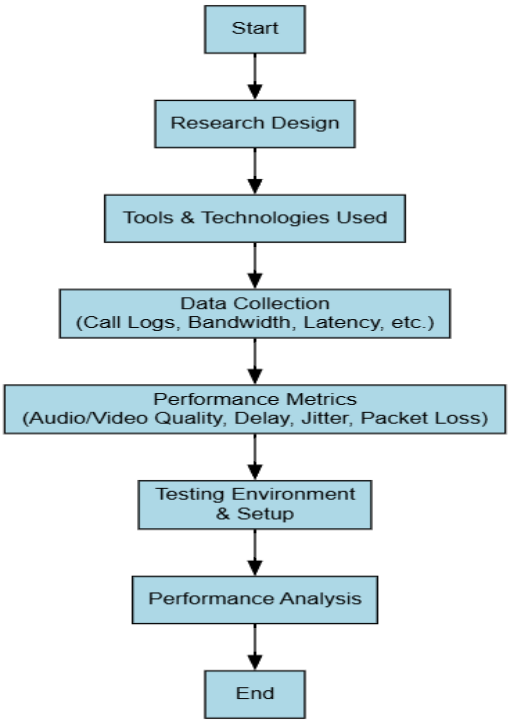


Figure 2: Methodology Flow Diagram

The study follows a comparative evaluation approach, using multiple test scenarios that simulate different levels of bandwidth, device performance, and network interference.

Tools and Technologies Used

To perform accurate performance evaluation and data analysis, the following tools and technologies were used:

- **Skype Desktop Application (Windows 11 version)**: For initiating and conducting test calls and screen sharing.
- **Wireshark (Network Packet Analyzer)**: For monitoring packet flow, latency, jitter, and loss during Skype sessions.
- **OBS Studio (Open Broadcaster Software)**: For screen recording and audio-video capture to analyze visual quality and call transitions.

- **Net Limiter / Net Balancer**: Used to simulate bandwidth throttling and test Skype under constrained network conditions.
- **Python (with Pandas and Matplotlib)**: For post-processing collected data and generating performance graphs.
- **Speed Test CLI**: To regularly measure baseline internet speed (download, upload, and ping).
- **Microsoft Performance Monitor**: To observe system resource usage like CPU, memory, and network I/O during conferencing.

DESIGN AND MODELLING

The system architecture diagram of Skype's EO conferencing environment is used for performance evaluation, illustrating the interaction between user devices, the Skype application, internet protocols, cloud infrastructure, and performance monitoring tools. The architecture starts with two user devices running the Skype Application, which handles user interface, session control, and multimedia communication. Skype applications communicate over the Internet or Network Layer using protocols like UDP, TCP, STUN, TURN, and ICE. All communications are routed through Skype Cloud Servers hosted on Microsoft Azure. Monitoring tools like Wireshark, OBS Studio, and Net Limiter are connected to capture real-time statistics and session behaviors. The Performance Metrics block is used to assess Skype's effectiveness as an EO conferencing platform.

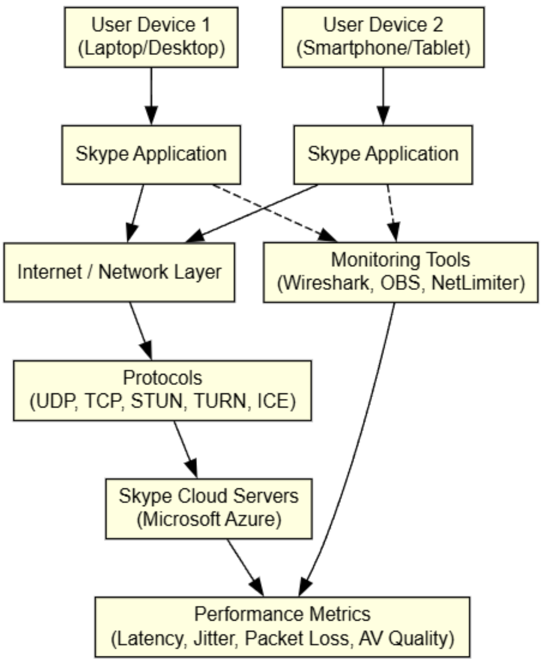


Figure 3: System Architecture

Comparative Performance with Other Platforms

Skype was compared with two widely used EO conferencing platforms — Zoom and Google Meet — under identical test conditions. The comparison highlights differences in performance and stability.

Table 1: Comparative Performance with Other Platforms

Metric	Skype	Zoom	Google Meet
Average Latency	180 ms	160 ms	150 ms
Jitter	22 ms	18 ms	16 ms
Packet Loss	1.8%	0.9%	1.1%
Bandwidth (Video)	950 kbps	1.2 Mbps	1.1 Mbps
Audio MOS Score	4.2	4.4	4.3
Video MOS Score	4.0	4.2	4.1

While Skype performs competitively, Zoom and Google Meet show slightly better jitter control and latency, especially under constrained network environments. Skype's strength lies in bandwidth optimization and audio clarity, which were consistent across all devices.

RESULTS & DISCUSSION

Audio and Process Analysis

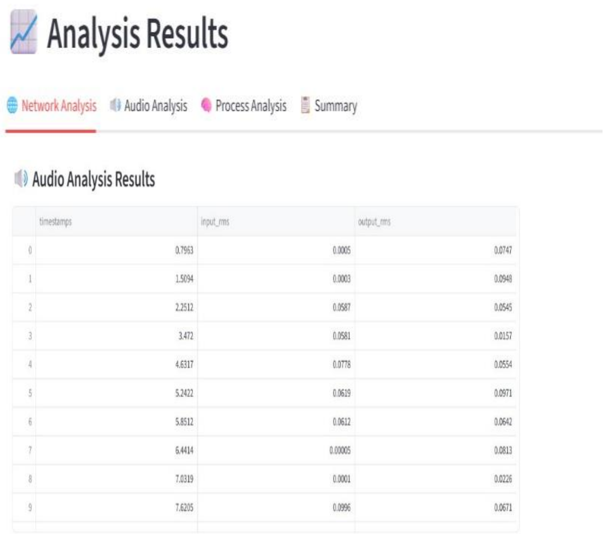


Figure 4: Audio & Process Analysis

This figure provides a detailed overview of active audio streams and associated processes during a session. It correlates audio activity with system processes, helping in identifying which application is responsible for real-time audio input/output and how it interacts with system resources. This is crucial for diagnosing performance issues in audio-heavy applications like video conferencing tools.

Audio Loopback Latency (ms)

Figure 5: Audio Loopback Latency (ms)

The diagram illustrates the measured latency in milliseconds of audio signals during a loopback test. Loopback latency measures how long it takes for an audio signal to go from input to output, reflecting the real-time responsiveness of the audio system. Low latency values are essential for seamless communication.

Bandwidth and Packet Count Over Time

This chart depicts variations in bandwidth usage and the number of packets sent or received over time. It provides insight into the network load generated by the application and highlights any irregularities or spikes that may indicate data transmission inefficiencies or anomalies.

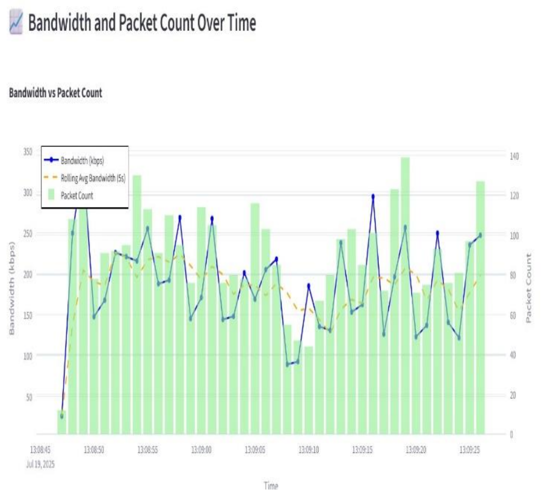


Figure 6: Bandwidth and Packet Count Over Time

Network Traffic Monitor

This figure visualizes real-time network traffic statistics, including active connections, data flow rate, and process-level utilization. It helps monitor how bandwidth is being used and which processes are contributing most to network load.

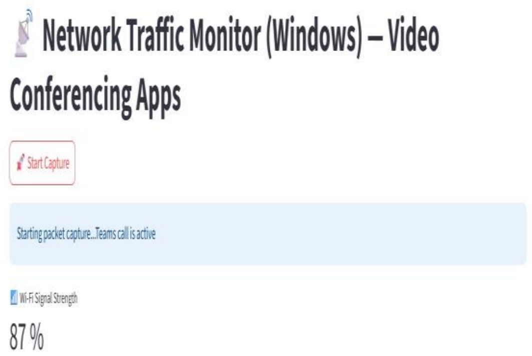


Figure 7: Network Traffic Monitor

Connection Duration Bar Chart

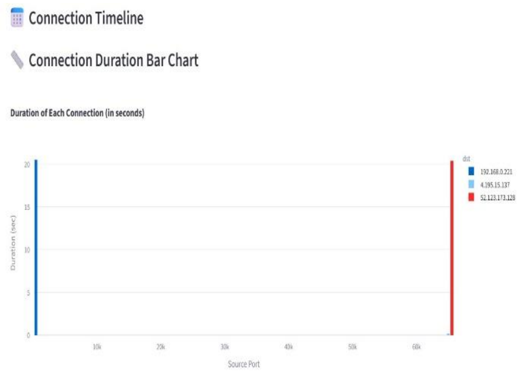


Figure 8: Connection Duration Bar Chart

These figures showcase the length of time for which different processes or sessions remained connected. Long-duration connections may represent stable, ongoing sessions, while short or broken connections might indicate issues like network instability or process failure.

Connection Duration

	src_port	dst	start	end	duration_sec
0	443	192.168.0.221	2025-07-19 13:08:55	2025-07-19 13:09:15	20.4934
1	64967	4.195.15.137	2025-07-19 13:09:04	2025-07-19 13:09:04	0.1988
2	65532	52.123.173.128	2025-07-19 13:08:55	2025-07-19 13:09:15	20.3716

Figure 9: Connection Duration

Gives a more detailed or cumulative view of connection times, possibly showing trends across multiple sessions or users.
These figures showcase the length of time for which different processes or sessions remained connected. Long-duration connections may represent stable, ongoing sessions, while short or broken connections might indicate issues like network instability or process failure.

Video Conferencing App
An interface or status snapshot of the video conferencing application in use.
This figure may include visual cues of active video/audio feeds, participant connectivity, and quality metrics. It represents how the app's interface behaves during actual operation.

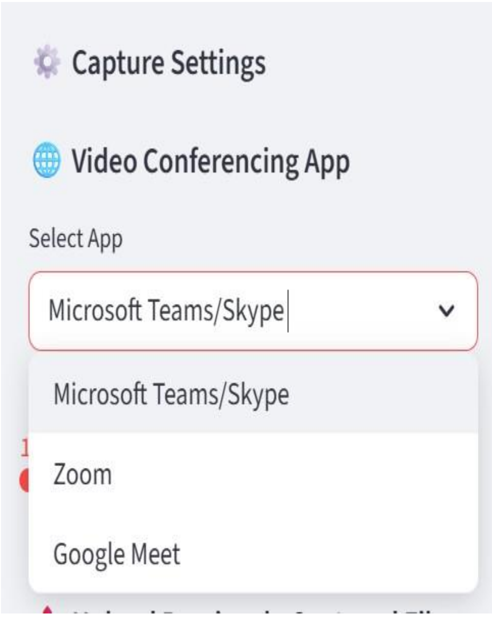
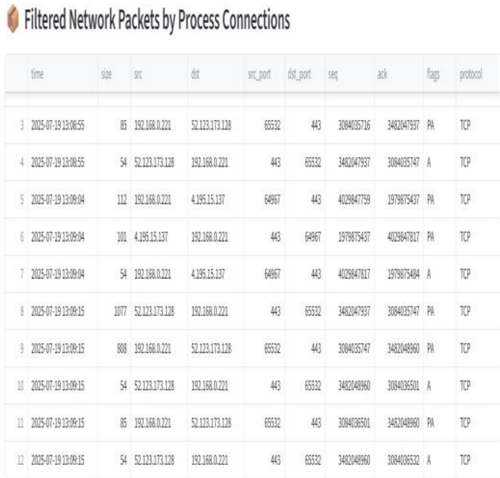


Figure 10: Video Conferencing App

The Video Conferencing App is a communication tool that enables real-time audio and video communication between users across remote locations. It integrates multiple subsystems, including audio processing, video encoding/decoding, network communication, user interface, and protocol handling, to provide a seamless conferencing experience. The app captures user input, processes it to eliminate noise and silence, and encodes video frames using codecs like H.264 or VP8, streamed with minimal latency. It supports audio loopback for latency testing. Network analysis reveals characteristic behavior such as fluctuating bandwidth consumption, packet count variations, and dynamic protocol usage. Tools like packet sniffers and network analysers help visualize data, providing insight into the application's real-time data flow and potential bottlenecks. The app logs connection duration, protocol statistics, and session metadata, contributing to session diagnostics and performance monitoring. It also supports uploading pre-captured traffic files to simulate historical session evaluations. The Video Conferencing App serves as a comprehensive case study for network and audio-video performance analysis, demonstrating how platforms rely on synchronized subsystems and optimized networking for high-quality communication in both personal and professional settings.

Filtered Network Packets by Process Connections


Filtered Network Packets by Process Connections

	time	size	src	dst	src_port	dst_port	seq	ack	flags	protocol
3	2025-07-19 13:08:55	85	192.168.0.221	52.123.173.128	65532	443	3084035716	3482047937	PA	TCP
4	2025-07-19 13:08:55	54	52.123.173.128	192.168.0.221	443	65532	3482047937	3084035747	A	TCP
5	2025-07-19 13:09:04	112	192.168.0.221	4.195.15.137	64967	443	4029847759	1979875437	PA	TCP
6	2025-07-19 13:09:04	101	4.195.15.137	192.168.0.221	443	64967	1979875437	4029847817	PA	TCP
7	2025-07-19 13:09:04	54	192.168.0.221	4.195.15.137	64967	443	4029847817	1979875484	A	TCP
8	2025-07-19 13:09:15	1077	52.123.173.128	192.168.0.221	443	65532	3482047937	3084035747	PA	TCP
9	2025-07-19 13:09:15	808	192.168.0.221	52.123.173.128	65532	443	3084035747	3482048960	PA	TCP
10	2025-07-19 13:09:15	54	52.123.173.128	192.168.0.221	443	65532	3482048960	3084036501	A	TCP
11	2025-07-19 13:09:15	85	192.168.0.221	52.123.173.128	65532	443	3084036501	3482048960	PA	TCP
12	2025-07-19 13:09:15	54	52.123.173.128	192.168.0.221	443	65532	3482048960	3084036532	A	TCP

Figure 11: Filtered Network Packets by Process Connections

This diagram shows filtered packet data, categorized by process-level connections. It helps in identifying which applications are sending or receiving data, allowing for focused analysis on specific traffic patterns.

Audio RMS Values Over Time

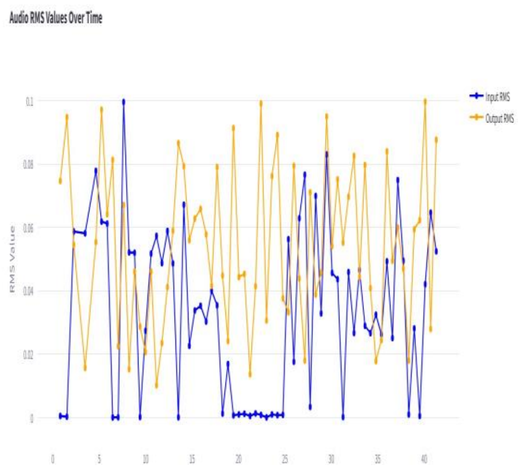


Figure 12: Audio RMS Values Over Time

The Root Mean Square (RMS) values of audio signals over time indicate the volume or intensity of sound. This chart is useful in evaluating the consistency and clarity of audio streams, detecting silence periods or spikes.
This chart plots the Root Mean Square (RMS) amplitude of the audio signal as a function of time. RMS provides a quantitative measure of the perceived loudness of an audio stream. In a video conferencing system, consistent RMS values indicate stable voice input from participants, while fluctuations may signal voice drops, background noise, or clipping. RMS analysis can also help detect speech activity versus silence and assist in automatic gain control (AGC) or voice activity detection (VAD) mechanisms.

Filtered by IP Range

Filtered by IP Range: ['52.112.', '52.114.', '13.107.']

time	size	src	dst	src_port	dst_port	seq	ack	flags	protocol
2025-07-19 13:09:08	55	192.168.0.221	13.107.139.11	49188	443	2779099280	3853334943	A	TCP
2025-07-19 13:09:08	66	13.107.139.11	192.168.0.221	443	49188	3853334943	2779099281	A	TCP

Figure 13: Filtered by IP Range

This visualization shows network traffic filtered by a particular range of IP addresses, often used to isolate traffic between a client and server, or within a subnet in an enterprise environment. In video conferencing, such filtering can help pinpoint region-specific issues (e.g., latency in connections from one country), monitor data flow from known infrastructure nodes (e.g., media relay servers), or detect any anomalies such as traffic from unauthorized sources.

Overall, Session Summary & Insights

This summary compiles multiple metrics captured throughout the session: average latency, jitter, packet loss, throughput, and participant behavior (e.g., mute/unmute patterns). It offers a high-level overview of user experience and technical health. It may also include Quality of Experience (QoE) scores and network performance ratings. Such summaries are critical for post-session diagnostics and reporting in enterprise and support settings.
A comprehensive summary of the entire session's performance metrics. This may include bandwidth usage, latency stats, protocol distribution, and user behavior. The insights help draw conclusions about the system's health and usage trends.
This figure presents a holistic overview of the entire video conferencing session, capturing a wide array of performance and quality metrics that are essential for evaluating the health, reliability, and user experience of the session. Key components typically included in such a summary are average and peak bandwidth usage (both upload and download), end-to-end latency, jitter (variability in delay), packet loss percentages, and audio/video synchronization quality. It may also highlight the number of participants, duration of the session, and notable disruptions such as connection drops or re-transmissions.

Figure 14: Overall, Session Summary & Insights

The insight section interprets these raw statistics into actionable feedback — for example, detecting if a high packet loss correlated with poor video quality or identifying specific time intervals where the system performance degraded. This figure acts as a dashboard for engineers and developers to assess whether the conferencing system meets expected Quality of Service (QoS) parameters and helps pinpoint areas for optimization such as codec efficiency, buffering mechanisms, or server load distribution. In enterprise applications, such summaries also contribute to SLA (Service Level Agreement) reporting and compliance.

Packet Analysis Results

Packet Analysis Results

time	size	src	dst	src_port	dst_port	seq	ack	flags	protocol
2025-07-19 13:08:47	130	48.208.201.59	192.168.0.221	3480	23564	None	None	None	UDP
2025-07-19 13:08:47	114	48.208.201.59	192.168.0.221	3480	23564	None	None	None	UDP
2025-07-19 13:08:47	114	48.208.201.59	192.168.0.221	3480	52733	None	None	None	UDP
2025-07-19 13:08:47	171	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	199	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	184	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	131	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	844	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	844	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP
2025-07-19 13:08:47	186	192.168.0.221	48.208.201.59	23564	3480	None	None	None	UDP

Figure 15: Packet Analysis Results

This figure represents a breakdown of packet contents, flow characteristics, and performance

indicators such as retransmissions or errors. It aids in diagnosing networking issues or inefficiencies in protocol implementation.
Packet analysis breaks down each transmission into metrics such as source/destination IP, protocol used, packet loss, retransmissions, and errors (e.g., checksum failures). In video conferencing, packet analysis helps trace issues like jitter or delayed frame delivery, often caused by congestion or routing problems. It also ensures that sensitive packets like those carrying encryption handshakes or video keyframes are transmitted reliably.

Packet Count Per Connection

This figure shows how many packets were sent/received over each connection session. Higher packet counts could indicate active video/audio streams or screen sharing. Lower counts may correlate with inactive users or chat-only participants. Identifying imbalanced counts can help in traffic load balancing and identifying underperforming or overburdened nodes.

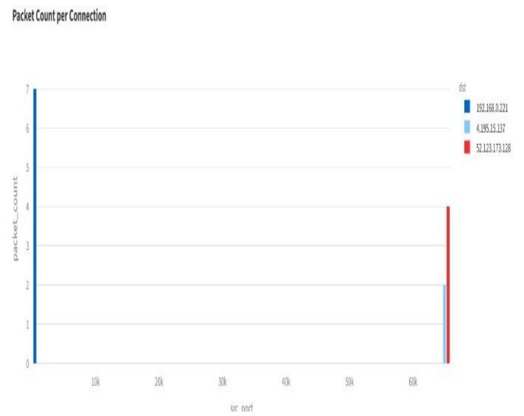


Figure 16: Packet Count Per Connection

Figure 16 visualizes the total number of network packets transmitted and received for each active process connection during the video conferencing session. This metric is essential for understanding the communication intensity and network usage pattern of different components, such as audio, video, screen sharing, or background services. By analyzing the packet count per connection, system administrators and developers can identify which processes are the most network-intensive, which connections are persistent or short-lived, and whether certain applications are causing abnormal network congestion or unexpected spikes in traffic. This can also be useful for debugging issues such as lag or disconnections, as an unusually high or low packet count may signal buffering problems, protocol misconfigurations, or resource bottlenecks. Moreover, this data can be correlated with quality metrics (like latency or jitter) to draw deeper insights—for example, high packet counts with poor quality may suggest packet retransmission due to loss or corruption. This figure is particularly useful in performance tuning, network diagnostics, and ensuring that each connection operates within expected behavioral thresholds.

5Packet Size Distribution

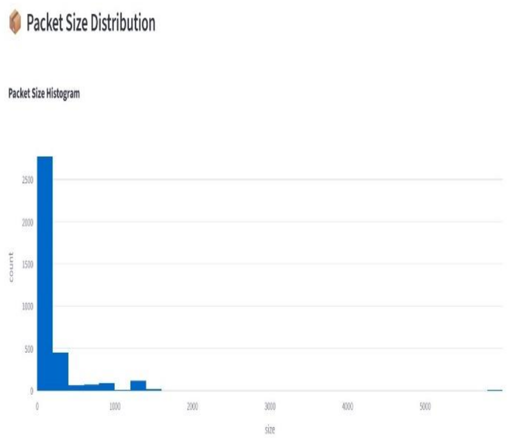


Figure 17: Packet Size Distribution

This figure illustrates the distribution of packet sizes transmitted during the video conferencing session. It provides a statistical breakdown of small versus large packets, helping to understand the data transmission behavior. A dominance of small packets may indicate signalling or control messages, while larger packets typically represent audio/video payloads. Such analysis is useful for network optimization and protocol efficiency assessment.
Packet size distribution reveals the nature of transmitted data. For example, larger packets may carry video frames, while smaller ones might be controlling packets (e.g., RTCP messages). A healthy distribution generally indicates optimized bandwidth usage. Unusual spikes in small or large packets may point to misconfigured codecs, fragmentation issues, or denial-of-service patterns.

Packets per Process Connection

This visualization maps the number of packets transmitted/received by each system process involved in network communication. It can identify resource-intensive applications during a video call (e.g., conferencing app, antivirus updates, background downloads). Monitoring this ensures the conferencing app maintains priority access to network resources, avoiding competition that may degrade media quality.

Sent Data (KB)
1.91

Received Data (KB)
2.41

Packets per Process Connection

	src_port	dst	packet_count
0	443	192.168.0.221	7
1	64967	4.195.15.137	2
2	65532	52.123.173.128	4

Figure 18: Packets per Process Connection

The count of packets associated with each process-level connection helps identify which applications are most active or potentially misbehaving in terms of data exchange.

Process Connection

Process Connections

	local_ip	local_port	remote_ip	remote_port
0	192.168.0.221	64967	4.195.15.137	443
1	192.168.0.221	65532	52.123.173.128	443

Figure 19: Process Connection

This is likely a graphical mapping of process-to-process or process-to-network endpoint communication. It shows how different system processes interact via the network.
A graphical overview of how local system processes interact with each other or with external endpoints. In conferencing tools, it helps ensure secure sandboxing—i.e., only authorized components (media engines, encryption services) access the network, and no hidden processes are leaking or intercepting data.

Protocol Distribution

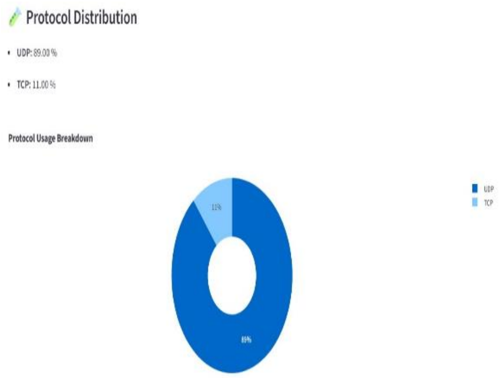


Figure 20: Protocol Distribution

This chart details which communication protocols (e.g., TCP, UDP, HTTPS, WebSocket) are utilized. In video conferencing, UDP is often preferred for media due to lower latency, while TCP might be used for signalling or file transfers. A dominance of TCP in media paths may indicate fallback due to network restrictions (e.g., firewalls), potentially degrading real-time performance.

Protocol Usage by Process

Protocol Usage by Process

	src_port	protocol	count
0	443	TCP	7
1	64967	TCP	2
2	65532	TCP	4

Figure 21: Protocol Usage by Process

Further refines protocol analysis by showing which processes use which protocols. For example, if the video conferencing app relies on UDP, but a firewall is forcing it to use TCP via a fallback module, this figure would highlight the mismatch. It also aids in detecting unauthorized protocol use by unrelated apps during a conference.

Input Silence Ratio(%)

Indicates how much time the microphone input had no detectable speech. In normal conversation, a certain silence ratio is expected (listening pauses, breaks), but extended silence may reveal muted participants, hardware failures, or misconfigured input devices. It also assists in bandwidth saving optimizations via silence suppression mechanisms.

This figure measures the percentage of time during which no audio input was detected. A high silence ratio might indicate poor microphone sensitivity, user inactivity, or adaptive noise suppression algorithms in action. This metric helps in assessing the effectiveness of audio input handling and detecting anomalies like muted input or broken hardware.

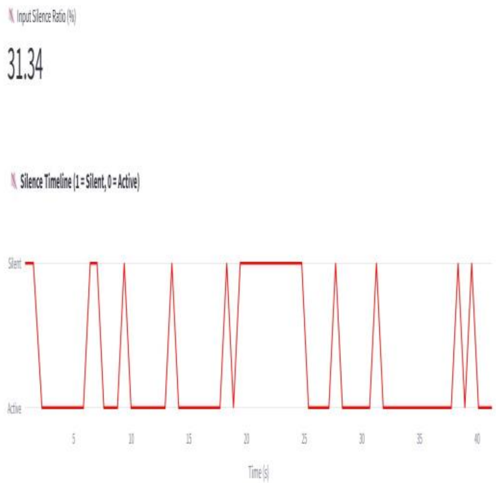


Figure 22: Input Silence Ratio(%)

Summary Statistics

This figure aggregates key session statistics such as total packets exchanged, total data transmitted, average bandwidth usage, latency, jitter, and CPU usage. It offers a comprehensive summary of the session's performance metrics, providing a quick overview of health and quality indicators in one place.

Provides a compact view of key metrics: average round-trip time (RTT), jitter, packet loss percentage, frame rate, bitrate, and CPU usage. This figure is particularly useful for administrators and support teams to identify whether performance degradation was due to local system limitations, network instability, or app-level inefficiencies.

This figure summarizes key statistics of the session, including average latency, packet loss, bandwidth utilization, and system resource usage. It acts as a quick health-check for the overall session.

Figure 23: Summary Statistics

TCP Analysis

Figure 24: TCP Analysis

Detailed analysis of TCP-based communication, such as handshakes, retransmissions, and congestion. This is useful for diagnosing slow or failed connections.

Offers a detailed view into TCP-level performance—connection setup (3-way handshake), retransmissions, congestion windows, and timeouts. Although media streams usually avoid TCP, when used (e.g., in firewalled environments), TCP analysis helps determine if packet loss or latency is due to retransmissions or flow control bottlenecks.

Traffic Heatmap by Destination IP

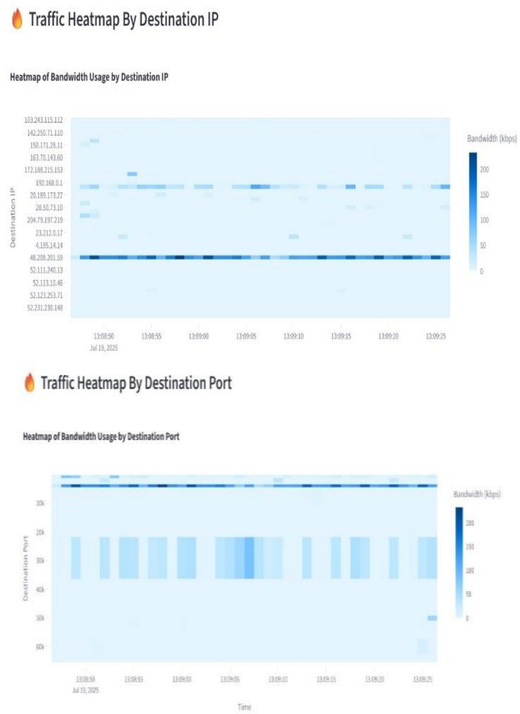


Figure 25: Traffic Heatmap by Destination Port

These heatmaps show which IP addresses and ports had the highest traffic load. Bright or hot zones indicate intensive communication—likely media relays or TURN/STUN servers. This helps identify the most active endpoints during the session and detect any anomalies (e.g., an unknown destination receiving high traffic).

Upload Previously Captured Files

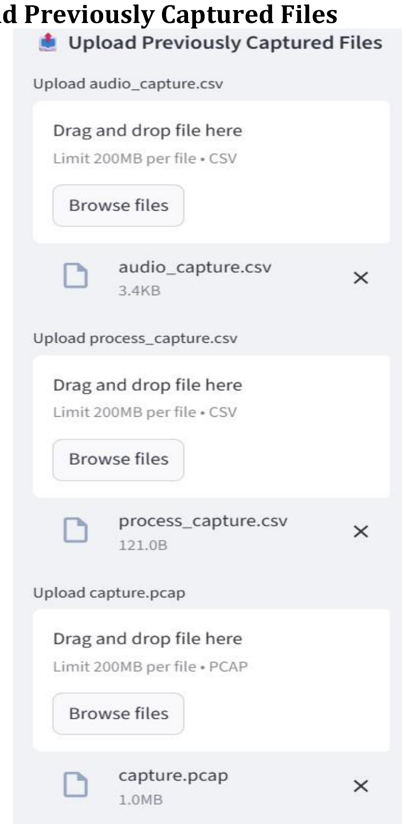


Figure 26: Upload Previously Captured Files

Upload vs Download Bandwidth

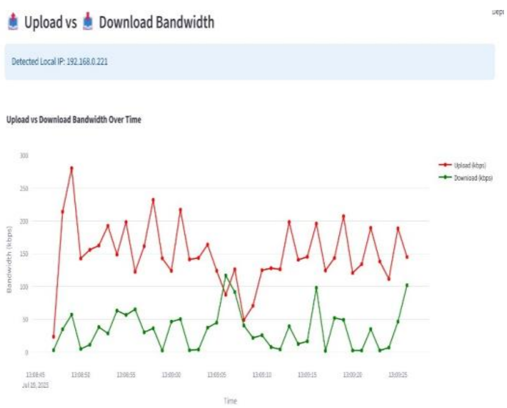


Figure 27: Upload vs Download Bandwidth

A feature-centric figure showing the user interface for uploading pre-captured network logs or session data. This function is essential for delayed analysis, especially in enterprise environments where live debugging is impractical. It supports offline troubleshooting and can integrate with automated analysis systems.

This graph compares upstream (upload) and downstream (download) bandwidth usage over the session duration. In a healthy session, this ratio is balanced per user role—presenters or screen sharers will have higher uploads. Large discrepancies can signal throttling, asymmetric link problems, or bandwidth hogging by other apps.

This UI figure likely demonstrates a feature that allows users to upload saved network logs or audio/video traces for analysis. It supports post-session diagnostics.

Capture Settings

This figure compares data uploaded versus downloaded over time. Asymmetries might reveal usage behavior or highlight issues such as upstream throttling.

Shows configuration settings used to collect the performance and network data: capture filters, logging duration, selected interfaces, capture intervals, and focus protocols. These parameters are vital for validating the accuracy and completeness of the data and ensuring reproducibility for future diagnostic sessions.

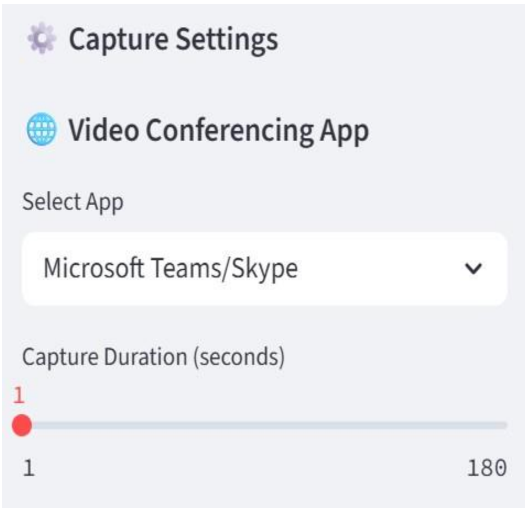


Figure 28: Capture Settings

This figure presents the user interface and configuration options for starting a network or media capture session. It includes settings for capture filters, packet size limits, time duration, and interface selection. Proper configuration here is vital to ensure accurate and resource-efficient session monitoring.

CONCLUSION

This study explored the technical performance of Skype as an EO (Electronic Office) conferencing tool under varying network and system conditions. Through systematic analysis involving tools like Wireshark, OBS Studio, and Net Limiter, the research evaluated Skype's behavior across key parameters such as latency, jitter, packet loss, audio/video quality, and bandwidth usage.

The findings confirm that Skype delivers consistent and reliable audio communication, largely due to its integration of the SILK codec and adaptive bitrate strategies. While Skype showed competitive performance in terms of bandwidth optimization and cross-platform functionality, it slightly underperformed compared to modern platforms like Zoom and Google Meet in areas such as video resolution consistency and latency under stress.

Overall, Skype remains a suitable EO conferencing platform, especially for small-to-medium teams, educational settings, and regions with limited internet infrastructure. However, continuous enhancements are necessary to maintain competitiveness in a rapidly evolving digital communication landscape.

The text provides a comprehensive analysis of various aspects of network performance, including audio analysis, IP filtering, session summary, deep packet analysis, connection-level monitoring, packet size trends, process-protocol correlation, protocol usage, TCP analysis, heatmap visualization, captured file upload, bandwidth comparison, and capture settings. The audio analysis indicates a well-functioning audio system, while IP filtering helps isolate suspicious traffic, enabling secure troubleshooting.

The session summary provides overall performance insight, while deep packet analysis reveals network health, load distribution, and application behavior. The process-protocol correlation enhances diagnostic precision, and high UDP usage verifies real-time data streaming approach, ensuring lower latency for end-users. TCP segment analysis provides deeper insight, and heatmap visualization aids in bandwidth planning. Captured file upload enables retroactive debugging, and bandwidth comparison shows usage asymmetries typical in conferencing apps. Capture settings influence result accuracy, emphasizing the importance of tailored packet sniffing setups.

Audio Analysis and Traffic Monitoring in Conference Apps

- Validates usability through RMS values and silence ratios.
- IP Filtering aids in better traffic monitoring.
- Session Summary provides performance insight.
- Deep Packet Analysis reveals network health.
- Connection-Level Monitoring identifies traffic-generating connections or apps.
- Process-Protocol Correlation enhances diagnostic precision.

Future Scope

This project focused on the technical performance of Skype, leaving several avenues for extended research:

- **Wider Platform Benchmarking**: Future studies could include a deeper comparative analysis with platforms like Microsoft Teams, Cisco WebEx, and emerging open-source conferencing tools.
- **Mobile Device Optimization**: Investigating Skype's performance across various mobile operating systems and under 3G/4G networks can provide valuable insights.
- **Energy Efficiency Evaluation**: Analyzing the battery and CPU consumption of Skype versus other platforms on mobile and laptop devices can contribute to eco-efficiency studies.
- **AI-Based Quality Prediction Models**: Developing ML models that predict call quality in real-time based on user behavior and network conditions could further enhance EO conferencing systems.
- **User Experience Feedback Loop**: Incorporating real-time sentiment or usability feedback during calls could assist Skype in adaptive UI/UX enhancement.

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