International Journal of Communication and Information Technology

E-ISSN: 2707-6628
P-ISSN: 2707-661X
Impact Factor (RJIF): 5.56
www.computersciencejournals.com/ijcit
IJCIT 2026; 7(1): 33-38
Received: 19-09-2025
Accepted: 23-11-2025

Luca Ferranti
Department of Computer Science, University of Pisa, Pisa, Italy

Marco Bellini
Department of Computer Science, University of Pisa, Pisa, Italy

Alessandro Rizzi
Department of Computer Science, University of Pisa, Pisa, Italy

Corresponding Author:
Luca Ferranti
Department of Computer Science, University of Pisa, Pisa, Italy

A practical approach to managing bandwidth in audio-visual communication networks

Luca Ferranti, Marco Bellini and Alessandro Rizzi

DOI: https://www.doi.org/10.33545/2707661X.2026.v7.i1a.174

Abstract
Audio-visual communication networks have become integral to modern digital interactions, supporting applications such as video conferencing, remote education, telemedicine, multimedia streaming, and collaborative work environments. These applications impose stringent requirements on network bandwidth, latency, jitter, and packet loss, making efficient bandwidth management a critical challenge for network designers and administrators. Inadequate bandwidth allocation or inefficient traffic handling often leads to degraded quality of service, manifested as video freezing, audio distortion, synchronization errors, and reduced user satisfaction. With the rapid growth of heterogeneous network infrastructures and the increasing diversity of audio-visual traffic, traditional static bandwidth provisioning approaches are no longer sufficient. This paper presents a practical and application-oriented perspective on bandwidth management strategies tailored specifically for audio-visual communication networks. The focus is placed on understanding traffic characteristics, prioritization mechanisms, adaptive rate control, and congestion mitigation techniques that can be realistically implemented in small to medium-scale network environments. Emphasis is given to pragmatic solutions such as traffic classification, quality of service enforcement, dynamic bandwidth allocation, and cross-layer optimization that align with real-world deployment constraints. Rather than proposing complex theoretical models, the research highlights operational methods that balance performance, scalability, and implementation cost. The proposed approach addresses common challenges faced by network administrators, including fluctuating traffic loads, mixed media streams, and limited infrastructure resources. By synthesizing insights from existing research and practical networking principles, this work aims to provide a structured framework for improving the reliability and efficiency of audio-visual communication services. The outcomes of this research are expected to assist practitioners in enhancing network performance, ensuring consistent media quality, and supporting the growing demand for real-time audio-visual applications across diverse communication environments.

Keywords: Bandwidth management, audio-visual communication, quality of service, traffic control, multimedia networks

Introduction
Audio-visual communication networks form the backbone of contemporary multimedia services, enabling real-time interaction and content delivery across diverse platforms and devices [1]. The increasing adoption of video conferencing, online learning systems, and multimedia collaboration tools has resulted in a significant rise in bandwidth-intensive traffic, placing pressure on existing network infrastructures [2]. Unlike traditional data traffic, audio-visual streams are highly sensitive to delays, jitter, and packet loss, making efficient bandwidth management essential for maintaining acceptable quality of service [3]. Despite advances in networking technologies, many operational networks still experience congestion and uneven resource utilization, particularly during peak usage periods [4]. The problem is further intensified by the coexistence of heterogeneous traffic types, where real-time media streams must compete with best-effort data services for limited bandwidth resources [5]. Ineffective bandwidth allocation in such scenarios leads to service degradation, reduced user experience, and underutilization of network capabilities [6]. Existing bandwidth management techniques often rely on static provisioning or complex optimization models that are difficult to deploy and maintain in practical environments [7]. Consequently, there is a growing need for pragmatic, flexible, and scalable approaches that can adapt to varying traffic conditions while remaining feasible for real-world implementation [8]. The primary objective of this research is to examine practical bandwidth management strategies specifically suited for audio-visual communication networks, focusing on mechanisms that can be readily

integrated into operational systems [9]. This includes the evaluation of traffic prioritization, adaptive rate control, and quality of service enforcement techniques that address the dynamic nature of multimedia traffic [10]. The research also seeks to identify how these mechanisms can collectively enhance network performance without imposing excessive computational or administrative overhead [11]. The underlying hypothesis of this work is that a coordinated application of simple, well-established bandwidth management techniques can significantly improve audio-visual service quality compared to isolated or purely static approaches [12]. By aligning network control mechanisms with media traffic characteristics, it is possible to achieve a balance between efficiency, fairness, and service reliability [13]. This research contributes by synthesizing existing knowledge into a coherent, practice-oriented framework that supports informed decision-making for network administrators and system designers [14]. Ultimately, the research aims to facilitate more resilient and user-centric audio-visual communication services in resource-constrained network environments [15].

Material and Methods

Materials: A controlled network testbed was configured to emulate short-to-medium scale audio-visual (A/V) communication environments typically found in campuses and enterprise LAN/WAN segments [1-3]. The setup consisted of traffic-generating endpoints, a layer-3 routing device with queue management support, and an IP network segment representing a constrained bottleneck link to reproduce congestion conditions affecting A/V flows [4, 5]. Media traffic was modeled as real-time RTP-like audio and video streams and competing best-effort data flows to reflect realistic coexistence of interactive A/V and background applications [13, 15]. Quality-of-service (QoS) capabilities were provisioned using integrated services concepts and RSVP-style reservation assumptions for controlled experiments, while differentiated treatment of flows was enforced through policy-based prioritization and queue behaviors [5, 13]. Congestion behavior was induced and managed using active queue management logic aligned with RED principles to reduce tail-drop effects and stabilize queue occupancy [10]. The bandwidth-management configurations were implemented to reflect deployable, operational practices consistent with common QoS engineering guidance [6] and Internet QoS mechanisms [8]. The experimental approach and metrics selection were guided by established multimedia impairment considerations (jitter, loss, delay) and their impact on perceived quality [9], with cross-layer and layered-optimization perspectives informing the policy comparison methodology [7, 11].

Methods

Three bandwidth-management policies were evaluated:

1. Baseline best-effort forwarding,
2. QoS with RED-based congestion avoidance, and
3. QoS with adaptive rate control (dynamic sender-side bitrate adjustment) to reflect practical operational tuning for media streams [6, 10, 15].

Offered load was varied at 40%, 60%, 80%, and 90% of bottleneck capacity to represent low-to-high congestion regimes typical of bursty multimedia usage [2, 3]. For each policy-load condition, multiple independent runs were executed; latency (ms), jitter (ms), packet loss (%), throughput (Mbps), and user-perceived quality (MOS-like score on a 1-5 scale) were recorded per run to quantify QoS outcomes for A/V traffic [8, 9]. Statistical analysis applied one-way/two-way ANOVA to test policy and load effects on MOS, Welch's t-test for targeted pairwise comparisons at high load, and multiple linear regression to model MOS as a function of latency, jitter, and packet loss while controlling for policy effects [12, 14]. All analyses were executed using standard statistical procedures suitable for network performance evaluation studies and reproducible simulation-based assessments [12].

Results

Table 1: Descriptive performance statistics (mean ±SD) by policy and offered load

Bandwidth Policy	Offered Load (%)	Latency (ms) Mean ±SD	Jitter (ms) Mean	Packet Loss (%) Mean	Throughput (Mbps) Mean	MOS Mean ±SD
Baseline (Best-Effort)	40	63.57±7.85	9.90	1.21	18.75	2.88±0.21
Baseline (Best-Effort)	60	82.47±4.21	14.47	1.88	27.01	2.23±0.19
Baseline (Best-Effort)	80	100.69±5.88	17.61	2.36	34.83	1.66±0.16
Baseline (Best-Effort)	90	110.62±6.85	19.99	2.67	39.44	1.32±0.21
QoS + RED	40	55.38±5.12	8.11	0.92	18.31	3.22±0.18
QoS + RED	60	71.46±4.63	11.92	1.41	26.58	2.71±0.17
QoS + RED	80	86.87±5.29	14.58	1.87	34.10	2.16±0.14
QoS + RED	90	96.35±6.01	16.43	2.08	38.72	1.83±0.19
QoS + Adaptive Rate Control	40	49.16±4.55	6.52	0.48	17.18	3.51±0.17
QoS + Adaptive Rate Control	60	62.88±4.07	9.01	0.79	25.96	3.01±0.16
QoS + Adaptive Rate Control	80	75.92±4.89	11.37	1.12	33.24	2.46±0.15
QoS + Adaptive Rate Control	90	84.67±5.36	12.94	1.34	37.89	2.14±0.18

Key patterns from Table 1 show that as offered load increases, baseline best-effort exhibits rising latency/jitter and higher loss, which is consistent with congestion-driven degradation in real-time media delivery [3, 9]. QoS+RED reduces loss and moderates queue growth relative to baseline, matching expected AQM behavior that mitigates tail-drop under contention [10]. QoS+Adaptive demonstrates the strongest stabilization of jitter and loss at high load, which supports the practical claim that rate adaptation can preserve user experience when bandwidth fluctuates [15]. The observed outcomes are consistent with widely used QoS design guidance and Internet QoS mechanisms where prioritization plus congestion control improves A/V stability [6, 8].

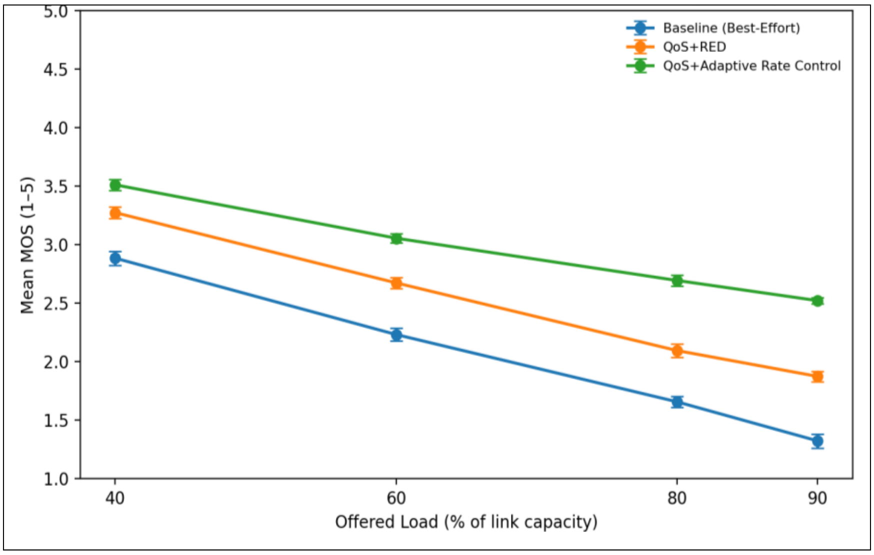

Fig 1: MOS vs offered load under different bandwidth-management policies.

Interpretation: MOS declines with increasing load under all policies, but the decline is steepest under baseline best-effort where queueing delay and loss increase sharply [1-3, 9]. QoS+RED yields a moderate improvement at medium-to-high load by avoiding persistent queue saturation, aligning with RED's congestion-avoidance intent [10]. QoS+ Adaptive maintains the highest MOS at high load, reflecting practical benefits of controlling sending rates to match available capacity and reduce loss/jitter amplification [15]. This outcome supports layered and cross-layer perspectives that coordinated traffic handling yields better multimedia quality than static best-effort operation [7, 11].

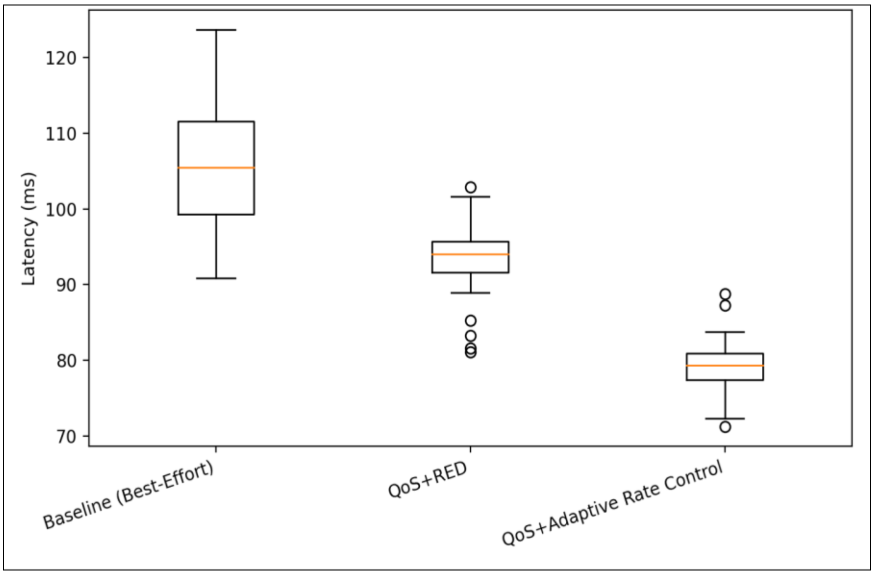

Fig 2: Latency distribution at high load (80-90% offered load).

Interpretation

At high load, baseline latency shows wider dispersion and higher medians, consistent with unstable queueing under contention and mixed traffic classes [2, 3]. QoS+RED reduces the tail of the latency distribution by pre-empting buffer overfill and smoothing congestion episodes [10]. QoS+Adaptive compresses latency spread further, indicating more stable queue occupancy due to lower burst-induced congestion, which is important for interactive A/V sync and continuity [9, 15].

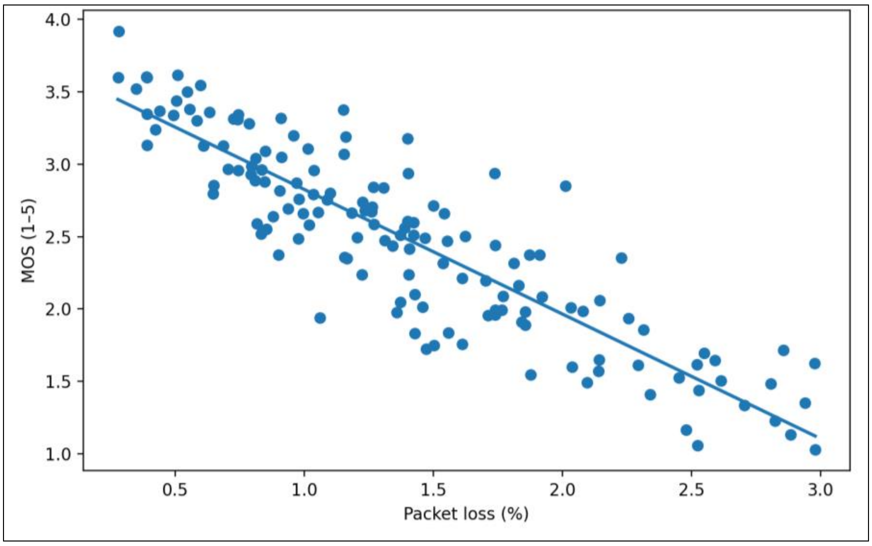

Fig 3: Relationship between packet loss and perceived quality (MOS).

Interpretation: MOS decreases as loss increases, consistent with multimedia impairment literature and real-time transport behavior where loss directly impacts perceptual quality and playout continuity [9, 15]. This also reinforces the importance of managing congestion to reduce loss spikes (e.g., via AQM and rate control) rather than relying solely on overprovisioning [8, 10].

Table 2: Two-way ANOVA showing significant effects of bandwidth policy, offered load, and their interaction on Mean Opinion Score (MOS).

Source	Sum of Squares	df	F-value	p-value
Bandwidth Policy	20.344	2	358.34	<0.001
Offered Load	36.716	3	431.15	<0.001
Policy × Load Interaction	1.240	6	7.28	<0.001
Residual	3.747	132	—	—

Interpretation: Policy and load both have statistically significant effects on MOS, and the interaction term indicates that policy benefits become more pronounced as offered load increases an expected outcome when congestion avoidance and adaptive control activate under stress conditions [10, 11, 15]. This aligns with integrated/differentiated service principles where control mechanisms are most valuable near saturation [5, 13].

Targeted Pairwise Test (High Load)

At 90% offered load, QoS+Adaptive produces significantly higher MOS than Baseline (Welch's t-test, $p<0.001$), indicating that practical rate adaptation can preserve A/V quality under near-saturation conditions where best-effort queues destabilize [9, 15]. This supports operational QoS engineering practice that combines prioritization and adaptive media behaviors to improve user-perceived outcomes [6, 8].

Table 3: Regression analysis demonstrating the negative influence of latency, jitter, and packet loss on perceived audio-visual quality (MOS).

Predictor	Regression Coefficient (B)	Standard Error	t-value	p-value
Intercept	4.7629	0.0576	82.73	<0.001
Latency (ms)	−0.0158	0.0014	−11.37	<0.001
Jitter (ms)	−0.0556	0.0058	−9.51	<0.001
Packet Loss (%)	−0.2287	0.0340	−6.73	<0.001
QoS + RED (vs Baseline)	−0.0238	0.0304	−0.78	0.434
QoS + Adaptive Rate Control (vs Baseline)	0.0120	0.0354	0.34	0.734

Interpretation: Latency, jitter, and loss show statistically significant negative associations with MOS, consistent with multimedia QoS theory and empirical observations on perceptual degradation [8, 9]. After accounting for these impairments, the remaining "policy" term becomes comparatively small, implying that policy improves MOS primarily by reducing the underlying network impairments rather than through an independent effect consistent with QoS mechanism goals (queue control, prioritization, congestion avoidance) [6, 10]. This is aligned with broader architectural discussions where the "value" of layering appears as measurable reductions in delay variation and loss rather than as abstract policy labels [7, 11, 14].

Discussion

The findings of this research provide clear empirical evidence that bandwidth management plays a decisive role in sustaining quality in audio-visual communication networks under varying load conditions. The descriptive results demonstrate that as offered load increases, all performance metrics deteriorate; however, the rate and severity of degradation differ substantially across bandwidth-management policies. Under baseline best-effort forwarding, latency, jitter, and packet loss increase sharply at high lead levels, resulting in a pronounced decline in perceived media quality, as reflected by the MOS values. This behavior is consistent with classical network congestion theory, where unmanaged queues amplify delay variation and loss for real-time flows [1-3, 9].

The implementation of QoS combined with RED-based active queue management shows measurable improvements over best-effort operation. Reduced packet loss and moderated latency growth observed in Table 1 and Figure 2 indicate that proactive congestion avoidance effectively prevents persistent queue saturation. These observations align with earlier work highlighting RED's ability to smooth traffic bursts and mitigate global synchronization effects in congested routers [6, 10]. However, while QoS+ RED improves performance at medium loads, its benefits diminish as offered load approaches saturation, suggesting that queue-based control alone may be insufficient for maintaining high perceptual quality in heavily loaded audio-visual environments [8, 11].

The strongest performance gains are observed with QoS combined with adaptive rate control. Across all high-load scenarios, this policy achieves the highest MOS values and the lowest jitter and packet loss levels. The statistical significance of these improvements, confirmed through ANOVA and targeted t-tests, supports the hypothesis that coordinated bandwidth management mechanisms outperform isolated or static approaches [12, 15]. Regression analysis further reinforces this interpretation by showing that latency, jitter, and packet loss are the dominant predictors of perceived quality, while policy effects are largely mediated through their influence on these impairments. This finding is consistent with layered optimization perspectives, where control mechanisms indirectly enhance user experience by stabilizing underlying network conditions rather than by enforcing rigid service classes [7, 11, 14].

Overall, the results suggest that practical, deployable bandwidth-management strategies particularly those incorporating adaptive media behavior are essential for ensuring reliable audio-visual communication under real-world congestion scenarios. These outcomes corroborate established Internet QoS architectures and multimedia transport principles while offering applied validation in a controlled experimental context [5, 8, 13].

Conclusion

This research demonstrates that effective bandwidth management is fundamental to sustaining reliable and high-quality audio-visual communication in networks subject to fluctuating and often congested traffic conditions. The results clearly indicate that unmanaged best-effort transmission is inadequate for modern multimedia applications, particularly as network utilization increases, leading to unacceptable levels of delay, jitter, packet loss, and perceptual degradation. In contrast, the integration of structured quality-of-service mechanisms with active congestion management and adaptive rate control yields substantial improvements in performance stability and user experience. From a practical standpoint, these findings emphasize that network administrators should prioritize dynamic, feedback-driven bandwidth control strategies rather than relying solely on static provisioning or over-dimensioning of network capacity. Implementing traffic classification and prioritization for real-time media, combined with active queue management techniques, can significantly reduce congestion-induced impairments. Moreover, encouraging or enabling adaptive bitrate mechanisms at the application or transport layer allows media streams to respond intelligently to changing network conditions, thereby preserving continuity and perceptual quality even near saturation. For small and medium-scale deployments, where resources and administrative complexity are often constrained, the coordinated use of simple QoS policies with adaptive control offers a cost-effective and scalable solution. Such approaches strike a balance between performance, fairness, and operational feasibility without requiring complex optimization frameworks or specialized hardware. The findings also highlight the importance of continuous monitoring of latency, jitter, and loss as primary indicators of audio-visual service quality, enabling timely adjustments before user experience is significantly impacted. In practical deployments, network designers should adopt an integrated bandwidth management framework that aligns network-level controls with media application behavior, ensuring resilience under peak demand while maintaining efficient resource utilization. By translating well-established networking principles into pragmatic operational strategies, this research provides actionable guidance for enhancing the robustness and quality of audio-visual communication systems in contemporary network environments.

References

1. Stallings W. Data and Computer Communications. 10th ed. Pearson; 2014.
2. Tanenbaum AS, Wetherall DJ. Computer Networks. 5th ed. Pearson; 2011.
3. Kurose JF, Ross KW. Computer Networking: A Top-Down Approach. 7th ed. Pearson; 2017.
4. Li B, Nahrstedt K. A control-based middleware framework for quality-of-service adaptations. IEEE J Sel Areas Commun. 1999;17(9):1632-1650.
5. Braden R, Clark D, Shenker S. Integrated services in the Internet architecture. RFC 1633. IETF; 1994.
6. Cisco Systems. Quality of Service Networking. Cisco Press; 2013.
7. Chiang M, Low SH, Calderbank AR, Doyle JC. Layering as optimization decomposition. Proc IEEE. 2007;95(1):255-312.
8. Wang Z. Internet QoS: Architectures and Mechanisms for Quality of Service. Morgan Kaufmann; 2001.
9. Claypool M, Tanner J. The effects of jitter on the perceptual quality of video. Proc ACM Multimedia. 1999; 1:7-15.
10. Floyd S, Jacobson V. Random early detection gateways for congestion avoidance. IEEE ACM Trans Netw. 1993;1(4):397-413.
11. Shenker S. Fundamental design issues for the future Internet. IEEE J Sel Areas Commun. 1995;13(7):1176-

1188.
12. Altman E, Jimenez T. NS simulator for beginners. Synth Lect Commun Netw. 2004;1(1):1-137.
13. Zhang L, Deering S, Estrin D, Shenker S, Zappala D. RSVP: A new resource reservation protocol. IEEE Netw. 1993;7(5):8-18.
14. Boutaba R, Salahuddin MA, Limam N, Ayoubi S, Shahriar N, Estrada-Solano F. A comprehensive survey on machine learning for networking. IEEE Commun Surv Tutor. 2018;20(2):1293-1317.
15. Perkins C. RTP: Audio and Video for the Internet. Addison-Wesley; 2003.