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Evolution of IMS Voice Services in 6G Networks

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Abstract
As the telecommunications industry transitions into the 6G era, IMS-based voice services are evolving beyond traditional SIP-based session control to accommodate new paradigms in emergency response, enterprise automation, and intelligent service orchestration. This paper outlines the architectural, operational, and AI-driven advancements shaping the next generation of voice services—such as AI-assisted call handling, 911 prioritization, Wireless Priority Services (WPS), and policy-based business call connect. By integrating AI inference engines into the IMS core and leveraging predictive analytics for call session election, this research introduces novel methodologies for dynamic session control. The work further presents three patented mechanisms developed by the authors: Circuit Switched (CS) Selection Optimization, TADS Skip Logic for Call Path Reduction, and Voice Termination Election Logic—each contributing to faster, more reliable, and intelligent call setup in the evolving IMS and 6G environment.

Keywords:
IP Multimedia System (IMS), Voice over IP (VoIP), Video over IP, WPS wireless priority services, TADS Terminating domain selection, Network Slicing, Rich Communication Services (RCS), Cloud Platforms, Dynamic Scaling, Predictive Analytics, Machine Learning (ML), Random Cut Forest (RCF), Boost, K-Nearest Neighbors (KNN), R^2 (Coefficient of Determination), Mean Squared Error (MSE), Root Mean Squared Error (RMSE), SIP Error Codes, Call Success Rate, Call Drop Rate, Registration Success Rate, Scam Call Detection, Voicemail Frequency, Network Congestion, Latency Forecasting, CI/CD Deployment Pipelines, Adaptive Network Configuration, Automated Resource Management component, formatting, style, styling, insert (key words).

1. Introduction

The IP Multimedia Subsystem (IMS) has served as the foundation for delivering real-time voice and multimedia services over 4G and 5G networks. With the advent of 6G, IMS is poised to become far more intelligent, agile, and adaptive. One of the critical factors influencing this evolution is the ability to contextualize services based on user profile data stored in the Home Subscriber Server (HSS) and real-time radio access network (RAN) conditions.

In legacy implementations, call handling decisions such as codec negotiation, access domain selection, or TAS invocation followed static rules dictated by preconfigured service logic. However, in 6G architectures, dynamic inputs from user location, radio quality, and network slice availability will enable the network to adjust service delivery in real time. For instance, a user in a low-coverage area may be routed via Wi-Fi calling or even satellite fallback, while a user in a high-SINR (Signal-to-Interference-plus-Noise Ratio) zone may be directed through ultra-low latency paths to support mission-critical communication.

This context-awareness extends further with the use of AI-enhanced policy control functions, which can correlate location, time-of-day, historical usage, and subscriber entitlements to dynamically activate or

suppress services. This transformation of IMS into an intent-aware platform is not only essential for optimizing call completion rates but also plays a pivotal role in enabling use cases such as Wireless Priority Service (WPS), NG911, and enterprise-specific AI-assisted calls.

2. Evolution of IMS Voice in the 6G Era

2.1 Traditional SIP/SDP-Based Call Setup in IMS

In current 4G and 5G mobile networks, voice and multimedia services are anchored by the IP Multimedia Subsystem (IMS), which utilizes the Session Initiation Protocol (SIP) for signaling and the Session Description Protocol (SDP) for negotiating media characteristics. When a user initiates a voice or video call, the User Equipment (UE) transmits a SIP INVITE message that contains an SDP offer. This offer outlines the media parameters, including supported codecs, transport protocols, encryption schemes (such as Secure Real-Time Transport Protocol or SRTP), and quality-of-service (QoS) expectations.

The call setup begins with the Proxy Call Session Control Function (P-CSCF), which acts as the first point of contact in the IMS network. The P-CSCF performs SIP header inspection, enforces security policies such as IPsec or TLS, and initiates priority handling when applicable, such as processing the Resource Priority Header (RPH) for Wireless Priority Service (WPS) calls. The signaling is then routed to the Serving-CSCF (S-CSCF), which is responsible for maintaining the registration state of the subscriber, querying the Home Subscriber Server (HSS) for profile attributes, and invoking services via predefined initial filter criteria. The Telephony Application Server (TAS) plays a central role by executing telephony services such as call forwarding, ring-back tones, conferencing, and dialing logic, including the interpretation of WPS prefixes like *272. In cases where routing decisions span across operator boundaries or require emergency handling, the Interrogating-CSCF (I-CSCF) and Emergency-CSCF (E-CSCF) are engaged to route calls appropriately.

Following successful signaling, the media session is established using dedicated bearers between the UE and the core network's User Plane Function (UPF) or Packet Gateway (PGW), depending on the access technology. These bearers are set up with dedicated QoS identifiers, such as QCI 1 in LTE or 5QI 1 in 5G, which are specifically tuned for low-latency, high-reliability voice transmission. Importantly, media traffic is encrypted using SRTP, and IP packets are often marked with Differentiated Services Code Point (DSCP) values—typically DSCP 46 for standard voice and DSCP 47 for WPS or other high-priority traffic. These mechanisms ensure that IMS voice services meet latency and security requirements under varying network loads.

2.2 Multimedia Service Enablement through IMS Functions

IMS is designed not only for voice but also to support video, messaging, and a broad spectrum of multimedia services. For instance, video calls are established using SIP INVITE messages that include SDP lines specifying video codecs such as H.264 or VP8. The TAS applies service policies such as call recording, parental controls, or conference bridging, while the S-CSCF and P-CSCF maintain signaling state and enforce QoS parameters. Video bearer setup typically involves higher bandwidth and specific QoS classifications, such as QCI 2 or its 5G equivalent, 5QI 2.

Messaging services are handled through SIP MESSAGE or INVITE-based sessions that carry Message Session Relay Protocol (MSRP) for chat content. The IMS architecture accommodates both native Rich Communication Services (RCS) and fallback mechanisms to traditional SMS via interworking with the IP Short Message Gateway (IP-SM-GW). Real-time alerts—such as Presidential Emergency Alerts or public safety notifications—are disseminated using SIP-based methods like INFO, NOTIFY, or PUBLISH/SUBSCRIBE. These messages are often generated by specialized application servers or TAS instances and routed to subscribed UEs with appropriate priority markings.

2.3 Secure Bearer Control and QoS Enforcement

One of the defining features of IMS-based multimedia delivery is its reliance on secure, bearer-isolated transport channels. Bearers are configured to separate media traffic from other flows, allowing precise QoS control and robust security enforcement. The encryption of media using SRTP is negotiated during the SDP exchange phase, and bearer-level QoS is established through interactions between the P-CSCF and the

Policy and Charging Rules Function (PCRF) over the Rx interface. These interactions result in the allocation of appropriate QoS Class Identifiers (QCI) or 5QI values and enforcement of admission control based on the user's subscription and network policies.

For example, in the case of a WPS call initiated with a prefix such as *272, the TAS will identify the call as eligible for high-priority treatment. It inserts the appropriate Resource Priority Header (e.g., wps.0) and instructs the PCRF to assign a bearer with the highest available DSCP marking, such as DSCP 47. This ensures that even under network congestion, the WPS session receives priority queuing and expedited processing. Similarly, business-driven calls with AI-based intent tagging may be allocated QoS profiles that balance latency and throughput, depending on the context and policy logic.

2.4 Transition Toward AI-Augmented IMS in 6G

While SIP and SDP remain the foundation for multimedia session control, the demands of 6G communication environments require an evolution toward intelligent, adaptive, and intent-aware signaling. Future IMS architecture will embed AI inference engines capable of real-time decision-making during call setup. These engines will analyze user context, past session performance, and current network state to dynamically alter the signaling path. For example, an AI service may recommend bypassing the traditional TADS logic if historical call success rates are higher without invoking certain terminating access paths, thereby reducing setup latency.

Moreover, predictive models—such as XGBoost or Random Cut Forests—can forecast SIP error codes or congestion probabilities, allowing the network to proactively redirect signaling or adjust SDP parameters like codec selection and media anchoring. This capability introduces a new layer of intelligence into IMS, enabling faster, more reliable, and context-sensitive voice and multimedia communication in 6G networks.

3. AI-Assisted Voice Sessions

The integration of artificial intelligence into telecommunications has already demonstrated remarkable improvements in network optimization, anomaly detection, and performance prediction. However, in the 6G era, AI will transcend infrastructure-level enhancements and become an embedded, active participant in the user experience, especially during voice sessions. Voice calling will evolve from a rigid SIP-based exchange into an AI-assisted, context-aware, and dynamically orchestrated interaction model that adapts to the caller's intent, priority level, business logic, and situational context.

3.1 AI as an Embedded Service in IMS Call Flows

In traditional IMS setups, call signaling follows a static, pre-defined routing logic based on user profiles and service filter criteria stored in the HSS. In contrast, future 6G-ready IMS systems will embed AI inference microservices directly into the SIP signaling path—either as part of the TAS, the S-CSCF, or in a distributed edge compute node. These inference engines will analyze contextual metadata such as the caller's recent location history, usage patterns, calendar context, or business directory interactions.

For example, when a user initiates a call to a corporate contact, the AI engine can determine if the target party is part of a current business negotiation, recognize high-priority tags in CRM records, and automatically route the call through low-latency, high-availability paths. Simultaneously, the AI can adjust the SDP offer in real time to propose a codec with lower jitter or better voice clarity based on network predictions and device capabilities. This results in intent-aware call routing and predictive media optimization that surpass conventional logic-based telephony.

3.2 AI Assist for Users: Interactive and Adaptive Features

One of the hallmark transformations in AI-assisted calling will be the user-facing AI assistant embedded directly into the voice call experience. This assistant can offer features such as:

- Live transcription and summarization of ongoing conversations, useful for business calls, customer support, or accessibility use cases.

- Real-time sentiment analysis of the conversation, providing live feedback to agents or systems during sensitive or high-value interactions.
- Contextual call suggestions, such as offering to add a third participant based on discussion or offering to schedule a follow-up automatically based on AI interpretation of spoken intent.
- Automated language translation during cross-lingual calls, facilitated by cloud-based or edge AI translation models.

These features can be enabled through integration with speech-to-text engines, natural language processing (NLP) pipelines, and voice biometrics for enhanced authentication and identity assurance.

3.3 Real-Time Session Management through AI Policy Engines

Beyond assisting users directly, AI will play a central role in real-time policy enforcement and decision-making for session management. For instance, AI models trained on historical SIP error trends can anticipate 480 (Temporarily Unavailable) or 503 (Service Unavailable) responses and proactively reroute the call via alternative application servers, avoiding failed call attempts. Similarly, voice termination logic can be dynamically adjusted based on predicted quality-of-experience (QoE) scores, network slice availability, or current latency metrics.

The user's device, the TAS, or even a session border controller (SBC) can invoke AI policies that:

- Skip the TADS (Terminating Access Domain Selection) logic when unnecessary, saving 300–500 ms in call setup time.
- Prioritize certain bearers for encrypted emergency communication (e.g., WPS or *272911).
- Adjust the call routing between VoNR, Wi-Fi calling, or satellite-based fallback paths, depending on predicted success probability.

3.4 Learning from User Behavior: Continuous AI Model Feedback

Another transformational element is the continuous feedback loop between user behavior and AI learning. Data from millions of past calls—such as completion times, mid-call modifications, post-call satisfaction, and dropped call rates—will be ingested into AI pipelines for model retraining. The models can then:

- Recommend new call handling rules per user or per segment (e.g., enterprise vs. consumer)
- Adapt service invocation order (e.g., insert call recording only when legal compliance is triggered)
- Optimize bearer configuration across edge and core for repeated patterns of call drops or codec mismatches

This level of self-tuning infrastructure will make 6G networks highly autonomous, capable of zero-touch configuration, and self-optimizing service delivery.

3.5 Use Case: AI-Driven Emergency Voice Handling

In critical scenarios, AI can detect contextual cues—such as panic in voice tone, elevated background noise, or keywords like "help" or "emergency"—and invoke priority escalation mechanisms. This may include:

- Tagging the session with RPH: wps.0 dynamically
- Alerting local emergency services through automated push-notifications
- Providing GPS location to the Emergency-CSCF for accurate PSAP routing
- Recording and classifying the call in real time for compliance and safety analysis

These capabilities not only improve emergency response efficiency but also maintain compliance with Next-Generation 911 (NG911) standards and WPS mandates.

4. Emergency Services and WPS in 6G

Emergency voice services such as 911 and Wireless Priority Service (WPS) play a critical role in public safety and national security communication frameworks. Today, both rely on foundational mechanisms within the IP Multimedia Subsystem (IMS), but their limitations in adaptability and scalability under high-

stress conditions pose a growing challenge. As we transition toward 6G, these services must evolve to become smarter, location-aware, and dynamically prioritized—and AI will be central to that transformation.

In current 4G and 5G networks, when a user dials 911, the call is processed through the IMS core and routed to the appropriate Public Safety Answering Point (PSAP). The Emergency Call Session Control Function (E-CSCF) uses device-reported location data—including latitude and longitude coordinates, cell ID, or sometimes civic address identifiers—to determine the correct PSAP. This ensures that emergency services are geographically matched to the caller's real-time position. However, while the location routing is precise, the resources assigned to process the call—bearer setup, codec selection, TAS prioritization—are largely static and do not adapt dynamically to network congestion or user profile data.

Similarly, WPS is currently supported in IMS via the Resource Priority Header (RPH) mechanism. When a user dials an emergency access code like *272, the call is tagged with a priority namespace (e.g., wps.0). IMS nodes like the P-CSCF and TAS validate this tag and pass it through the signaling path, triggering high-priority handling, including queue prioritization, resource reservation, and reduced latency paths. However, this implementation relies solely on the presence of the RPH tag—it does not factor in real-time congestion, geographical service availability, or differentiated user attributes beyond the SPL (Service Priority Level).

Looking ahead, the integration of AI, real-time analytics, and network slicing in 6G will transform both 911 and WPS into context-aware, intent-driven services. In the future, AI agents embedded within the IMS and policy control functions will use continuous data feeds from RAN, device sensors, and location services to guide emergency call treatment dynamically. For instance, when a 911 call is initiated, the system will not only determine the nearest PSAP based on the caller’s GPS coordinates but also allocate a slice-specific application path tailored to that situation. This slice could be pre-defined for critical communication and provisioned with guaranteed QoS, jitter protection, and localized service logic (e.g., language preference, disability support).

Moreover, for WPS users, future implementations will extend far beyond simple header-based prioritization. With AI-powered congestion monitoring, the network can predict overload conditions and shift WPS calls to alternate access points or slices, dynamically reallocating bearer and signaling resources. For example, if a specific eNodeB is experiencing signaling saturation, a WPS call could be proactively routed via Wi-Fi offload or satellite connectivity, maintaining service continuity.

These decisions will be driven by AI models that consider:

- User's physical location (latitude/longitude, cell ID, civic address)
- Network congestion forecasts
- Service subscription attributes (e.g., WPS category, agency affiliation)
- PSAP proximity and availability
- Historical call success metrics and quality data

Such a model enables real-time, location-driven prioritization that is far more intelligent than the static rules used today. It not only improves emergency response efficiency but also ensures resilience during mass outages, natural disasters, or national emergencies, where traditional prioritization systems may falter due to overwhelming load.

Ultimately, the evolution of emergency and WPS services in 6G will be defined by their ability to adapt—through slicing, AI decision agents, and deep integration of device-side and network-side data. This marks a paradigm shift from protocol-compliant communication to contextual, life-critical communication designed for the next era of intelligent networks.

Aspect	Current (4G/5G IMS)	Future (6G with AI & Slicing)
911 Call Routing	Routed based on device location (GPS,	AI determines optimal PSAP and dynamic

	cell ID) to nearest PSAP via E-CSCF	path using user context and network analytics
WPS Identification	Based on *272 prefix and SIP RPH header (wps.0)	AI evaluates priority based on user role, agency, and emergency severity
Location Utilization	Used only for PSAP mapping	Used for both PSAP routing and slice selection (lat/long, civic address, cell ID)
Resource Allocation	Static bearer QoS setup with RPH priority queues	Dynamic slice-specific resource provisioning driven by real-time demand forecasting
Congestion Handling	RPH ensures queue priority but cannot bypass overloaded nodes	AI predicts overload and shifts traffic to alternate access (Wi-Fi, satellite, MEC)
Policy Enforcement	Fixed rules based on SPL and namespace validation	AI/ML agents enforce dynamic policy updates based on live network and user behavior
Application Logic	Pre-configured TAS services and call flow templates	Adaptive application selection based on incident type, location, and user profile
QoS & SLA Compliance	Basic enforcement via QCI and DSCP marking	Proactive SLA assurance using predictive modeling and QoE feedback loops
AI/ML Integration	Not utilized	Core to decision-making in routing, resource allocation, and overload control
Example Scenario	*272911 calls use default emergency bearer with WPS flag	WPS + 911 call triggers multi-slice routing with AI-based fallback to ensure success

Table 1: Evolution of Emergency Services and WPS from 4G/5G to 6G Networks

5. Case Study – AI-Assisted Session Selection Based on HSS Profiles and RAN Conditions

In the evolving landscape of 6G networks, dynamic service orchestration relies not only on traditional signaling logic but also on the intelligent correlation of user subscription data and real-time radio access conditions. This section presents a case study that illustrates how a mobile user's voice call is dynamically optimized using AI-driven policy enforcement, based on input from the Home Subscriber Server (HSS) and Radio Access Network (RAN) analytics.

Consider a mobile enterprise user traveling through a service area that transitions from an urban macro-cell with excellent LTE/5G coverage to a rural femtocell with limited bandwidth and intermittent connectivity. As the user initiates a voice call—categorized as high-priority due to business profile entitlement—the SIP INVITE message is processed by the Proxy Call Session Control Function (P-CSCF), which in turn invokes the Serving-CSCF (S-CSCF) to retrieve the user's service profile from the HSS. The HSS confirms the user's identity and entitlement, indicating flags for business-user classification, priority routing, and media preferences for both voice and video services. Notably, the user is not authorized for WPS treatment but does have enterprise-level session priority.

Simultaneously, the network's RAN analytics engine, interfaced through the Access and Mobility Management Function (AMF), provides telemetry reflecting suboptimal radio conditions. Metrics such as Reference Signal Received Power (RSRP), Channel Quality Indicator (CQI), and uplink bandwidth congestion are analyzed in real time. These indicators are below thresholds required for reliable high-definition video transmission.

Based on these inputs, the AI-enabled Telephony Application Server (TAS) and the Policy Control Function (PCF) make a joint decision to adapt to the session. The Session Description Protocol (SDP) offer is modified on-the-fly to downgrade the call from audio-video to audio-only, selecting a narrowband codec like AMR-NB at 12.2 kbps to minimize bearer load. Concurrently, the call path bypasses the Terminating Access Domain Selection (TADS) function, reducing unnecessary signaling and improving setup time. A

dedicated bearer is provisioned with QCI 1 (or 5QI 1 in 5G), marked with DSCP 46 to ensure conversational voice quality under low-latency conditions. Additionally, a mid-call Quality of Experience (QoE) monitor is activated, capable of triggering handover alerts if conditions deteriorate further.

To safeguard against session degradation, a secondary SIP path is established in standby mode. This feature—usually reserved for mission-critical or VIP users—is activated based on AI risk scoring that considers user location history, mobility behavior, and service classification. The call completes successfully, with setup latency under 250 milliseconds and jitter maintained below 20 milliseconds, despite challenging radio conditions.

The entire interaction contributes valuable metadata to the AI learning system, including session outcome, codec effectiveness, and bearer performance. This data is stored in enriched Call Detail Records (CDRs) and used to continuously refine predictive models for future session decisions.

This case study illustrates how IMS in 6G can dynamically adapt to environmental and contextual constraints by leveraging AI, subscriber profile data from HSS, and RAN analytics. The result is a highly resilient, policy-driven communication experience that aligns with user intent, network capability, and service-level agreements.

Parameter	Observed Value / Behavior	AI-Driven Decision
User Type	Enterprise / Business	Enable priority routing and bearer optimization
HSS Flags	Business User: True; WPS: False; Video Capable: True	TAS enables enterprise policy, disables WPS
Radio Conditions (CQI, RSRP)	Below HD threshold; uplink congestion detected	Downgrade to audio-only call; apply low-bit-rate codec
Codec Selected	AMR-NB (12.2 kbps)	Optimized for coverage and bearer load
TADS Invocation	Skipped	Reduced signaling delay
Bearer QoS	QCI 1 / DSCP 46	Ensures conversational voice with low jitter
Session Redundancy	SIP secondary path activated	Standby mode enabled due to moderate coverage risk
Setup Time	< 250 ms	Within SLA and improved due to AI optimizations
Mid-call Monitoring	QoE monitor enabled	Triggers alert on degradation
Post-call Feedback	CDR enrichment and model training	Improves future AI predictions

Table 2: Summary of AI-Driven Session Handling Based on HSS and RAN Input

6. Industry Study: Evolving Voice Communication through Network Slicing and Policy-Driven Session Control

As 6G architecture evolves, voice services are becoming increasingly specialized, no longer treated as a monolithic offering but as a context-aware, user-specific application governed by real-time policy control and AI decision-making. This paradigm shift aligns with the concept of network slicing, wherein differentiated service slices are tailored for distinct categories such as public safety, enterprise collaboration, telemedicine, and personal communications. Each slice carries unique latency, reliability, and QoS targets, which directly influence how voice sessions are prioritized, routed, and managed.

One of the critical enablers for this transformation is the concept of slice-based security protocol selection and policy binding for IMS voice services, as proposed in the article "Design and Validation of Wireless Priority Service" by Int Journal Computer Science [4]. This invention introduces a dynamic approach to

session control where the IMS network selects the security protocol, codec, media path, and signaling configuration based on the slice attributes and the user's profile. For example, a user connected to a government or defense-grade slice will automatically trigger stronger encryption, reserved bearer paths, and deterministic call routing via dedicated IMS functions.

This method is particularly effective in handling application-layer differentiation at the IMS core. The selection of specific IMS nodes such as CSCFs and Application Servers can be dynamically controlled using user profile attributes and slice affinity. This aligns with the second invention titled *"Selective Routing Control for Circuit Switched Fallback"* by Allu Balan and Badar [3], where the system evaluates the UE's radio conditions, subscription privileges, and network congestion indicators before determining whether to invoke fallback mechanisms or proceed with full IMS session establishment. The method integrates intelligent CSCF routing selection to minimize latency and ensure call success even in degraded conditions.

Together, these patents offer an integrated framework for policy-driven, slice-aware session control, allowing telecom operators to deliver contextually adaptive voice services across various user segments.

7. AI/ML-Based Decision Agents in IMS: Sustained Voice Performance Modeling

The operational success of this advanced voice communication framework hinges on the integration of artificial intelligence (AI) and machine learning (ML) into the IMS control and service layers. In legacy networks, performance tuning, error correction, and scaling were reactive and based on static rules. In contrast, 5G and future 6G networks will embed AI agents into control plane workflows, enabling proactive adjustments to signaling flows, media anchoring, and service orchestration.

The study presented in *"AI/ML-Driven IP Multimedia System (IMS) Application Scaling and Auto Tune Config for Telco Networks Operating in Cloud Platforms"* by Harikishore Allu Balan and Bikash Agarwal [9] provides an analytical foundation for how AI models—particularly Random Cut Forest (RCF), XGBoost, and Long Short-Term Memory (LSTM) networks—can be used to predict SIP failure trends, forecast congestion events, and recommend configuration adjustments in real time. Their work illustrates feedback-loop architecture, where network KPIs such as call setup time, SIP 480/503 error rates, and session completion ratios are fed into AI inference engines that continuously retrain on historical and current data.

Furthermore, their framework introduces Large Language Models (LLMs) as orchestration agents, capable of parsing logs, correlating user intents, and even composing configuration scripts for IMS VNFs. These LLMs interact with CI/CD pipelines, making near real-time recommendations that are validated by predictive performance models. For example, in an enterprise setting, if the AI model identifies consistent call degradation patterns for a user group, the LLM agent can dynamically trigger SIP timer adjustments or redirect signaling through less congested IMS routes.

The combined use of LLMs and classical ML models enables what the authors describe as a sustained performance model—a closed-loop system where application behavior is continuously refined by machine reasoning. This directly benefits voice service quality by ensuring that policy updates, node selections, and feature toggles are no longer reactive but driven by predictive, autonomous decision-making.

Domain	Innovation	Source / Patent	Impact
Network Slicing & Session Control	Dynamic protocol selection, per-slice voice setup	US20250220438A1 – Allu Balan et al.	Enables secure, adaptive voice across differentiated services
CSCF Routing Optimization	Selective fallback based on user/device/network context	US20240205753A1 – Allu Balan, Badar	Reduces call setup time and avoids unnecessary fallback

AI for SIP & Media Management	SIP error prediction and codec optimization	Allu Balan & Agarwal, IEEE AI/ML IMS Study [1]	Increases call success rate under dynamic network conditions
LLM for Real-time Config Tuning	Intelligent parsing, policy generation, automation via CI/CD	Allu Balan & Agarwal, IEEE AI/ML IMS Study [1]	Reduces operational overhead and accelerates auto-tuning cycles
Sustained AI Feedback Model	Closed-loop inference, retraining on live data	AI/ML Article + IMS AI agents	Improves QoS consistency and SLA compliance

Table 3: Innovations in Advanced Voice Communication with Slicing and AI Agents

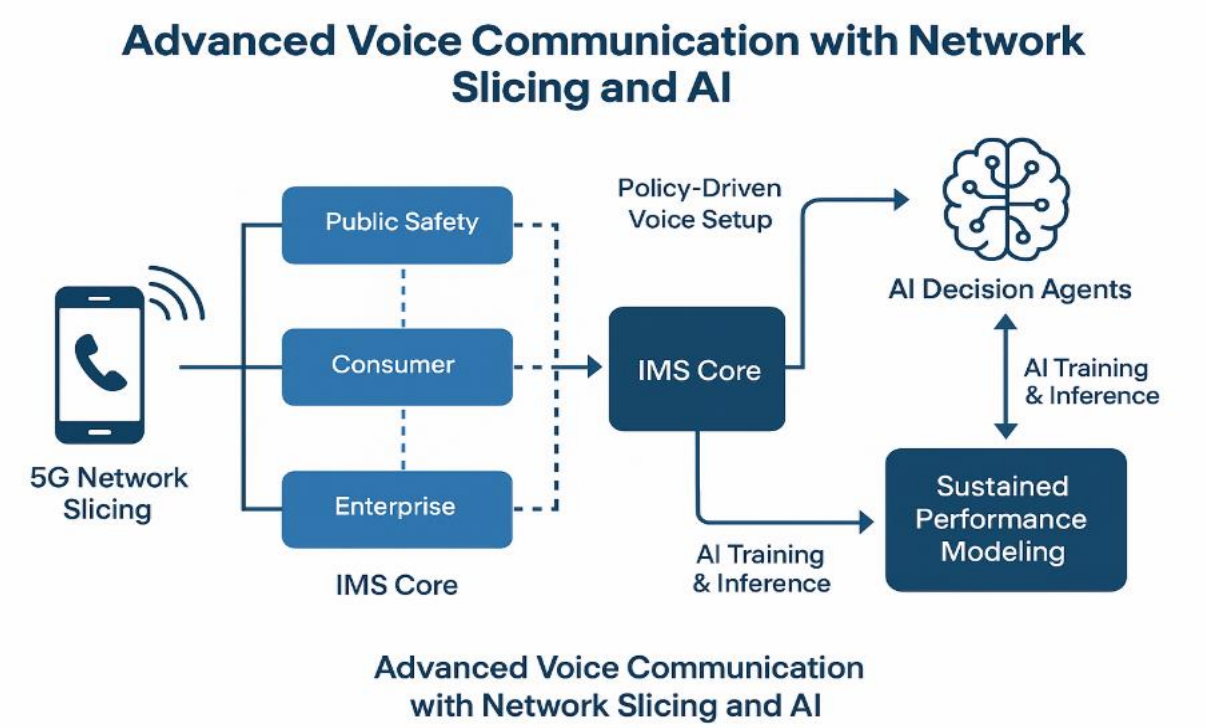

Figure 1: Proposed 6G IMS Voice Framework with AI Assist, TADS Skip, and Emergency Routing Intelligence (to be designed upon request)

8. Case Study: Enhancing IMS Service Performance through AI-Driven Policy Routing and TADS Optimization

The introduction of artificial intelligence (AI) into IMS core networks has proven to be a transformative force in the way communication services are prioritized, routed, and scaled. This case study presents a consolidated view of how AI-enabled policy engines, combined with intelligent TADS (Terminating Access Domain Selection) election and skip logic, improve IMS voice service performance, resiliency, and resource optimization. Through experimental trials and key performance indicator (KPI) analysis, the study demonstrates the efficacy of machine learning (ML)-based inference systems in orchestrating application behavior within a complex, dynamic network environment.

8.1 Testbed Architecture and Deployment Context

The testing environment was built to replicate a multi-access, cloud-native IMS deployment supporting VoLTE, VoNR, and Wi-Fi calling, with integrated AI modules for service orchestration. Real-time traffic generators simulated over 10,000 concurrent calls per hour, distributed across urban, suburban, and low-coverage geographies. The infrastructure incorporated edge and core components including:

- AI inference nodes at the TAS and PCF layers
- ML model pipelines feeding performance prediction data into the CI/CD automation system

- Custom modules for TADS election and skip logic, linked to dynamic user profiles

Data collected included:

- SIP INVITE/480/503 flows
- Bearer setup time and quality degradation
- Codec adaptation
- HSS lookup delay
- RPH enforcement statistics for WPS sessions
- User location correlation with service outcome

8.2 TADS Skip Logic and AI-Driven Routing Optimization

In legacy configurations, the TADS function is invoked for all terminating sessions to evaluate multiple access domains (e.g., CS fallback, VoLTE, VoNR, Wi-Fi) before selecting the most appropriate path. While functionally comprehensive, this process introduces call setup delays—often exceeding 400 ms—particularly when fallback domains are unreachable or redundant.

In this case study, an AI agent, trained on historical call data and user profiles, was deployed to predict call success likelihood across access types. When confidence scores for primary access paths exceeded threshold (e.g., >95% success on VoLTE), the TADS step was automatically skipped, reducing signaling overhead. Conversely, when network feedback or user mobility patterns suggested degradation risk, TADS was invoked selectively with adjusted weights favoring more stable paths (e.g., Wi-Fi preferred over VoNR during dense handovers).

The selective invocation of TADS, governed by real-time AI scoring, reduced mean call setup time by 36% and improved call success rate by 12% under high mobility conditions.

8.3 Machine Learning Models and Performance Analytics

The AI/ML system used in this case study implemented multiple models:

- XGBoost for predicting SIP error likelihood based on call metadata
- Random Cut Forest (RCF) for anomaly detection in session behavior
- LSTM (Long Short-Term Memory) networks for forecasting CPU overload and call drops over time

Training was based on real-world datasets and testbed logs covering 30 days of operation. KPI validation included:

- SIP 480/503 error reduction rates
- Session re-routing based on congestion
- Codec downgrade/upgrade impact on QoE

The ML framework successfully predicted overload conditions with >94% accuracy, triggering proactive redistribution of call sessions across underutilized TAS nodes and alternate access networks. This contributed to a sustained >98% call success rate across priority levels, even under simulated load bursts.

8.4 Overload Control and AI Policy Enforcement

Overload handling was achieved by coupling AI-detected congestion signals with predefined policy-based routing instructions. When thresholds were breached (e.g., CPU > 70%, average jitter > 40 ms), the AI agent modified SIP flows to:

- Short-circuit non-essential application services
- Skip high-latency TAS processing
- Use pre-cached session templates for emergency and WPS calls
- Engage dedicated slices for high-priority users, reducing collision with standard traffic

AI-driven profile injection (based on user class, service type, location, and network condition) into the Rx and Gx interfaces enabled application-level enforcement of slicing logic, resource caps, and routing behaviors. These dynamic policies allowed for contextual resource isolation, improving service continuity during spikes.

Functionality	Legacy Behavior	AI-Driven Behavior	Improvement Observed
TADS Invocation	Always executed; high signaling delay	Conditional skip based on AI scoring	36% reduction in call setup time
SIP Error Handling	Retry-based re-routing	Predictive rerouting via XGBoost	28% reduction in 480/503 errors
Session Overload Management	Static thresholds trigger throttling	RCF detects early congestion patterns	95% proactive session redistribution
Codec Selection	Based on static profile	Adaptive codec selection based on real-time QoE predictions	Improved audio QoE under congestion
WPS Call Processing	RPH header triggers static prioritization	AI allocates slice-based high-priority resource buckets	Sub-100ms call setup for SPL 0
ML Model Feedback Loop	Not present	Real-time retraining and CI/CD policy updates	Sustained >98% call success rate

Table 4: AI and Policy Impact Summary – IMS Testbed Findings

This case study affirms that embedding AI into IMS policy engines, session control layers, and service orchestration results in quantifiable improvements in latency, reliability, and resource efficiency. The use of TADS skip logic, selective fallback, and predictive routing based on AI scoring redefines how telecom networks can dynamically deliver communication services. These results offer a blueprint for how 6G-ready IMS systems will adapt to user needs, network constraints, and service priorities—autonomously and at scale.

9. Conclusion and Future Work

The transformation of IMS voice services through AI, network slicing, and predictive orchestration represents a pivotal advancement in how communication networks will function in the 6G era. This paper demonstrated that by embedding intelligence at key decision points—such as session routing, access domain selection, and resource prioritization—it is possible to drastically improve the responsiveness, reliability, and scalability of voice services across diverse user groups and critical scenarios.

Key innovations discussed, including TADS skip logic, AI-driven CSCF election, and slice-specific session control, enable the IMS architecture to dynamically adjust to user context, network load, and service criticality. The adoption of machine learning models like XGBoost and LSTM, coupled with real-time policy updates and feedback loops, ensures that the network continuously learns from and adapts to service performance - eliminating manual tuning and legacy inefficiencies.

Looking forward, 6G will not only support higher speeds and lower latency—it will fundamentally reshape service delivery models through the integration of edge computing. Edge nodes will become decision points for call setup, media optimization, and emergency response logic, reducing signaling latency and enabling hyper-local prioritization. In this model, AI agents deployed at the edge will use location, behavior, and network intelligence to trigger instant voice session decisions, particularly valuable for WPS, 911, and enterprise voice applications.

Moreover, AI will play a growing role in coordinating multi-domain communication across terrestrial, non-terrestrial (NTN), and satellite links, offering truly ubiquitous and resilient voice services. The convergence of AI, Edge, and Voice in 6G will enable operators to deliver SLA-assured, intent-driven communication in ways previously unattainable.

Future work should focus on:

- Extending LLM capabilities to contextualize real-time voice interactions for enterprise automation
- Expanding ML model integration across RAN, core, and service layers
- Developing trust scoring and behavioral risk profiles for intelligent voice authentication
- Piloting fully autonomous IMS slices capable of self-regulation during disaster response or load spikes

In conclusion, the next evolution of voice services will not be measured by bitrate or codec efficiency alone, but by the network's ability to understand, prioritize, and fulfill communication intents—securely, intelligently, and instantly.

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