Future of Voice Architecture

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1. INTRODUCTION AND SCOPE

Since telephony was invented more than 150 years ago, voice communication has become the primary communication method to connect everyone in the world. In the past century, telephony has evolved from analog to digital transmission, from circuit-switched to Internet Protocol (IP) packet-switched, and from wireline to wireless technology. Since the advent of cellular telephone networks in the late 1970s, wireless technology itself has evolved through five generations. In the past 20 years, since the original conception of IP Multimedia Subsystem (IMS), IMS has defined the architecture and protocols (e.g., Session Initiation Protocol (SIP)) for both wireline and wireless services around the world. In the transition from 4th-Generation (4G) wireless standards to 5th-Generation (5G) standards, the overall architecture evolved to a service-based architecture, but the core architecture and protocols for voice telephony services were little changed.

Voice has been increasingly expanding beyond human two-way communications into voice assistants, smart devices, gaming, and other machine-driven applications. The industry will need to explore new architectural approaches for efficiently managing voice and deliver a high-quality experience across a broad range of new applications.

The purpose of this report is to provide an architectural assessment for the next generation of voice services as the industry considers future architectural voice and data platforms in the 2030 timeframe. As voice continues to be more integrated into other multimedia applications, the industry will need to assess the future pathway of voice services related to consumer and enterprise markets.

2. CURRENT VOICE SERVICES

LANDSCAPE, STANDARD ACTIVITIES, AND CHALLENGES

2.1 Current Voice Services Landscape

Voice over Internet Protocol (VoIP) technology has replaced the limited-purpose end devices with software applications running on multi-purpose devices such as smartphones or laptop computers. Special-purpose circuit-switched access facilities have disappeared as the voice application can now utilize the packet-switched wireless or wireline IP access channel that supports all other IP-based applications to the devices. Meanwhile, the end-office switching systems have been replaced by application servers that reside in arbitrarily placed data centers that do not need to correspond to packet access network equipment locations. The transport networks between voice service providers are now interconnected IP cloud networks with appropriate isolation elements at the border (e.g., interconnection Session Border Controllers (SBCs)).

The replacement of circuit-switched end devices and switching systems with software applications and applications servers has also created the ability for service providers other than the facilities-based providers to offer voice services to users over the IP access channel. These service providers include VoIP providers that can deploy their own voice access application on subscriber end devices and deploy application servers in cloud data centers, utilizing the public internet as the transport between the two. Some VoIP providers offer a service interconnected with the public voice service provider networks, which primarily use telephone number-based addressing to reach other VoIP, mobile, and wireline users. Other VoIP providers offer business collaboration services that allow voice, messaging, conferencing, and other services within a subscriber organization or federated across organizations (via federation of directory information), as well as public voice network interconnection using telephone number addressing. Yet another set of providers integrate voice capabilities into social-network applications and infrastructure using their own user-name directories for addressing other users.

Some private and federated directory systems use a public E.164 telephone number as an identifier associated with users within their own directory or federation, independent of the number's use and routing on the public network. On the business application side, some providers offer communications services that can be integrated into other web-based business applications via Application Programming Interfaces (APIs) for handling contacts to or from users via a combination of voice and other applications implementing the customer's business processes. As the circuit-switched voice network disappears, the blurring of voice services with other communications, business, and social applications enabled by IP-based cloud applications is expected to become even more prevalent.

2.2 Standards Activities Related to Evolving Voice Architectures

2.2.1 3rd Generation Partnership Project (3GPP)

Most standards work related to voice services is taking place within 3GPP SA2 which address specific voice standards related to architectural aspects and IMS, as well as messaging services that use the same infrastructure as voice. Given the improvements in the radio over the past few generations, 3GPP SA4 is looking at a new voice codec that may be better suited for immersive voice. 3GPP SA6 considers application layer issues that may not be specific to voice.

SA2 Initiatives Defining Service-Based Architectures for IMS Interfaces

> SBA-based architectures leverage the advantages of Representational State Transfer (REST) APIs, including resource versioning, Hypertext Transfer Protocol (HTTP) security and caching, and enabling functions to more easily migrate to "cloud native."

> The 3GPP System Architecture (SA) and Core Terminals (CT) Working Groups have simplified the IMS core by defining a service-based architecture (SBA) option to replace the diameter interfaces to the Home Subscriber Server (HSS) and Policy Control Function (PCF).

> The 3GPP SA WG-2 "System Architecture and Services" is studying SBA-based approaches to discover and select IMS media functions.

SA2 Initiatives on IMS Architecture Enhancement

3GPP SA2 is developing a technical study TR 23.700-87 [1] that proposes IMS enhancements to enable the following new Real-Time Communication (RTC) services:

> Data Channel

> Augmented Reality (AR) communications

SA2 Initiatives on Message Service in 5G System

Short Message Service (SMS) was developed in 2nd-Generation (2G) systems and is still widely used today for both human and machine communications. With the introduction of 5G, support for Massive Internet of Things (MIoT) is a key market segment that requires enhancements to make messaging communications efficient. In addition, new 5G capabilities such as support of constrained devices and Non-IP Data Delivery (NIDD) could be leveraged to support new types of device endpoints. To evolve and expand the existing messaging service to fit into the 5G system, 3GPP started its Message service in 5G system (Msgin5G) work in Release 16 and completed its first phase of service enabler development in Release 17. This new service enabler covers not just the existing human-to-human messaging and machine-to-machine messaging, but also messaging between human and machine. An example of human-to-machine communication is people using voice or messages to control their home IoT devices from outside their house. An example of machine-to-human communication is home IoT device using messaging to communicate with its user in a pre-defined event is. Current MSGin5G service supports the following messages:

> Point-to-Point

> Application-to-Point

> Point-to-Application

> Group

> Broadcast

Enhancements of MSGin5G continue in 3GPP Release 18.

SA4 Initiatives on Voice Codecs

In networks prior to 5G, the coding and decoding latency inherent in complex codecs could be tolerated because the Radio Access Network (RAN) and Core Network (CN) themselves added large latencies. However, 4G, 5G, and, it is expected, 6G networks provide very low latency, very wide data bandwidth (even in relatively small spectrum spaces), and can provide very low error rates. Taking advantage of these radio enhancements opens the door to opportunities for new codecs that provide better conversational flow with reduced echo control requirements. An example is the immersive voice and audio service (IVAS) codec currently under consideration for Release 18 in 3GPP SA4 that provides optimized support for the future immersive voice services.

SA6 Initiatives to Enable New Vertical Applications

The 3GPP SA6 "Application Enablement and Critical Communication Applications" Working Group is tasked with providing application-layer frameworks, architectures, and mechanisms that simplify the deployment of northbound APIs to a variety of service verticals. Examples include Mission Critical (MC) services such as public safety, vehicle-to-everything (V2X) services, Factories-of-the-Future (FF) services and Uncrewed Aerial System (UAS) applications. SA6 has developed a number of general-purpose frameworks that simplify the development and deployment of public land mobile network (PLMN) and 3rd-party applications. Two of these frameworks — Common API Framework (CAPIF) and Service Enabler Architecture Layer for Verticals (SEAL) — provide functionality that could be applied to northbound APIs supporting real-time communication applications.

CAPIF TS 23.222 [2]

CAPIF provides a unified northbound API framework across the multiple 3GPP network functions supported by the PLMN (e.g., RAN, Packet Core, IMS). CAPIF supports a set of interfaces that provide application developers with a standard way to support functions that are common across all APIs. The functions supported by CAPIF include the following:

> Registration of new APIs.

> API discovery and enabling on-boarding/off-boarding of API invoker.

> Support for 3rd-party domains (e.g., to allow 3rd-party API providers to leverage CAPIF).

> Support for interconnection between two CAPIF providers.

> Federation of CAPIF functions to support distributed deployments.

> CAPIF events Subscription/Notification.

> Entity Authentication/Authorization.

> Secure communications.

SEAL TS 23.434 [3]

SEAL is similar to CAPIF in that it provides a standard way to support functions that are common across all APIs. However, SEAL focuses on functions related to an API's ability to access and control an individual user or group of users, including the following managements:

> Identity

> Configuration

> Location

> Group

> Key

> Network resource

2.2.2 Internet Engineering Task Force (IETF)

Just as current VoIP network architectures (SIP and IMS) are built using IETF developed standards, the Future Voice Architecture will draw upon existing and future IETF standards, as well.

One example of current IETF work (at the time of publication of this document) that may have a role in the Future Voice Architecture is the More Instant Messaging Interoperability (MIMI) IETF working group in the Applications and Real-Time Area. This working group's charter sets out the following goals:

“The More Instant Messaging Interoperability (MIMI) working group will specify the minimal set of mechanisms required to make modern Internet messaging services interoperable. Over time, messaging services have achieved widespread use, their feature sets have broadened, and their adoption of End-To-End Encryption (E2EE) has grown, but the lack of interoperability between these services continues to create a suboptimal user experience. The standards produced by the MIMI working group will allow for E2EE messaging services for both consumer and enterprise to interoperate without undermining the security guarantees that they provide. The working group will aim to achieve the strongest usable security and privacy properties for each targeted functional requirement.” [4]

3. THE VISION

The future voice network will seamlessly connect today's telecommunications world with the internet world and provide a true interoperable communication network beyond just voice and messaging. The future voice network will follow the basic principles of openness and decentralization and emphasize connecting everyone to ensure security, privacy, and accessibility. The future voice network will become the base infrastructure/underlying engine to provide signaling and control mechanisms to both basic voice services and more complex new services that may require extreme high bandwidth and real-time engineered communication services.

The future voice network will support service discovery, application capability discovery, network path capability discovery, and endpoint capability discovery. These will allow services to adapt to different access technologies or endpoints while still offering similar levels of integration and communication service support. It will allow service chaining and Network as Code to provide an end-to-end realization of services across those many communication layers and environments. In addition, the future voice network will support open interoperable interfaces toward applications, access, carriers, and endpoints. As such, it would take advantage of the feature-rich applications and the interoperability that the traditional telecommunications provider offers. This architecture would allow the adaptation to the fast-evolving applications landscape for voice and other communication.

In the future voice network, the user or endpoint may be reached by either traditional E.164 address or other unique web identifier (e.g., user@ domain type of address). The global identifier would be used to authenticate the user/endpoint in a decentralized fashion.

The same future voice network will be able to provide both the enhanced regulatory voice/video/mission critical services (e.g., Next Generation 911, Multimedia Priority Services (MPS), Communications Assistance for Law Enforcement Act (CALEA), and STIR/SHAKEN) and enhanced applications with enriched features such as Quality of Service (QoS) capability. The future voice network will support both a multi-layer, multiple provider's comprehensive network offering feature-rich service to end users and interworking to everyone with the QoS required for voice and other media.

The future voice network will be resilient and will play an important role in the future connected world. By collaborating with different players in the future communication ecosystem, future voice networks will help close the digital divide and provide a better future digital world for everyone.

4. FUTURE VOICE SERVICES

4.1 Overview

First and foremost, basic voice services will need to be carried forward. Extensions to voice that include video and AR/Virtual Reality (VR) also need to be carried forward or defined. Messaging and unified communications capabilities should be carried forward or defined, as well.

Regulatory voice services will continue to be supported. The latest communications technologies should be made available to the suite of Wireless and Multimedia Priority Services and emergency services — both for first responder groups and for the public interacting with the Public Safety Answering Points (PSAPs). Similarly ongoing support for CALEA will be critical to future voice services.

Although unified communications capabilities can be offered as part of a single solution, even more compelling is disaggregating these capabilities so that they can be provided as separate components via APIs for integration into any number of applications.

A. Voice

- Full-duplex voice
- Half-duplex voice (PTT)
- Voice messaging

B. Multimedia

- Video communications
- AR/VR
- Messaging
- Unified Communications-like capabilities
- Immersive Teleconferencing and Telepresence for Remote Terminals (ITT4RT)

These communications services will need to include offerings of supplementary features — traditional and new for both the consumer and business markets.

In addition to these features, APIs should be exposed via a Network as Code framework where useful network related capabilities (e.g., data connectivity, voice, messaging, network analytics, network data, User Experience (UE) data, edge cloud computing) are abstracted and simplified for consumption in application developer ecosystems. This enables any (3rd-party) application to integrate network awareness and control directly in the application code and with that, create altogether new types of applications and experiences.

In the future voice network, technical support services will increasingly leverage Artificial Intelligence (AI). Some examples of such services are listed below.

A. Technical support services

- Chatbots
- Remote, AI-assisted diagnostics and repair

B. Operations services

- Predictive maintenance
- Load balancing
- Cybersecurity
- Others

To facilitate the use of AI, a data access framework will allow data-consuming AI agents to obtain data from data-producing network components including from the 3GPP communications system (e.g., IMS).

The future will also see the introduction of new business models. For example, over the last couple of decades, advertising has been proven to be a successful business model for internet applications. This model can be applied to embedding advertisements in communications services. Flexibility to support other new business models will also be needed.

4.2 Government and Regulatory Services

4.2.1 Emergency Services

The evolution of Emergency Services in the U.S. is driven mainly by the following standards:

1. National Emergency Number Association (NENA) i3 standard for Next Generation 911, NENA-STA-010.3-2021 [5]
2. ATIS Standard for Implementation of 3GPP Common IMS Emergency Procedures for IMS Origination and ESInet/Legacy Selective Router Termination, ATIS-0700015.v005 [6]

NENA's family of NG9-1-1 standards, with i3 serving as the cornerstone, enables data-rich, secure, IP-based communications from the public, through 9-1-1, to every field responder. "i3" refers to the NG9-1-1 system architecture defined by NENA. It standardizes the structure and design of functional elements comprising the set of software services, databases, network elements, and interfaces needed to process multimedia emergency calls and data for NG9-1-1. The i3 solution supports E2E IP connectivity; gateways are used to accommodate legacy wireline and wireless originating networks that are non-IP, as well as PSAPs interconnected to the i3 solution architecture. NENA i3 introduces an IP-based inter-network (network of networks) called Emergency Services IP network (ESInet) with NG911 Core Services (NGCS) providing IP signaling and media for delivery of emergency calls to i3-capable PSAPs.

ATIS-0700015 defines the 3GPP IMS emergency procedures to support emergency call delivery to a NENA i3 ESInet or to a legacy selective router. It identifies the types of media that can be delivered to each type of emergency services network. The ATIS-0700015 abstract says:

"This document identifies and adapts as necessary 3GPP common IMS emergency procedures for applicability in North America to support emergency communications originating from an IMS subscriber (wireline or wireless; fixed, mobile or nomadic) and terminating at an ESInet, or, for appropriate media, legacy emergency services network to support Multimedia Emergency Services (MMES). It is the intent of this standard to support a full multimedia experience; therefore, simultaneous text, voice, pictures, and video are supported in this standard." [6]

4.2.2 CALEA

CALEA support will need to continue to evolve to support the surveillance of ever-expanding service offerings (e.g., voice, video, text, messaging) and network capabilities (e.g., STIR/SHAKEN). Key considerations for CALEA evolution include:

Convergence to 3GPP TS 33.128 Standard [7]

Historically, voice CALEA standards evolved as follows:

1. TIA J-STD-025 (A and B): Circuit-Switched Voice for Wireline and Wireless [8]
2. ATIS 1000678: Packet-Switched Voice (VoIP) for Wireline [9]
3. ATIS 0700005: Packet-Switched Voice (VoIP) for 3GPP IMS-Based VoIP and Multimedia Services (Wireline and Wireless) [10]

Going forward, 3GPP TS 33.128 [7] is the recommended roadmap target specification for voice and session-based services, due to having a roadmap for CALEA/LI coverage for new services and capabilities, such as:

- Rich Communications Services (RCS), in particular, HTTP-based File Transfer
- In-Bound Roaming (visiting PLMN (VPLMN) coverage) for Home-Based Routing (S8HR, N9HR)

ATIS may develop other specifications that build upon, or otherwise modify, the TS 33.128 target to support country-specific needs for lawful intercept.

3GPP currently supports operator lawful interception obligations and will continue to do so in the future.

4.2.3 Priority Service requirements

Several priority services have been developed for voice networks. In North America, Government Emergency Telecommunication Service (GETS) and Wireless Priority Service (WPS) support users making voice calls. Voice calls using these services are often collectively referred to as National Security or Emergency Preparedness calls (NS/EP calls) [11, 12, 13].

GETS was developed to support authorized government users' priority on wireline voice networks (e.g., public switch telephone network (PSTN)) during emergency situations. The goal was to increase the probability of the call connecting over a congested or partially failed network. Over time, additional services have been introduced (e.g., anonymity, VoIP access, PBX support) as use cases developed. GETS provides authorization and authentication via a PIN code entered by the user. Thus, the user is potentially able to use any wireline phone to receive GETS services. With the proliferation of wireless phones, a new service was required to ensure effective support for NS/EP calls.

WPS, or as it is referred to in 3GPP documents MPS, was originally developed to support the detection, authorization, and authentication of NS/EP calls on cellular networks, and how to grant these calls an increased probability of success during network congestion. Originally developed to support only voice calls, it has expanded the functionality to address video calls, text, and data services and is anticipated to extend to all communication services developed in the future. WPS supports five priority levels, with one being associated with the user's device according to their classification; from the highest level of government, through emergency responder coordinators and managers, down to volunteer agencies (e.g., Red Cross) and network service provider personnel responsible for network restoration. However, the service is not intended for use by first responders, who are expected to have communication devices other than cell phones.

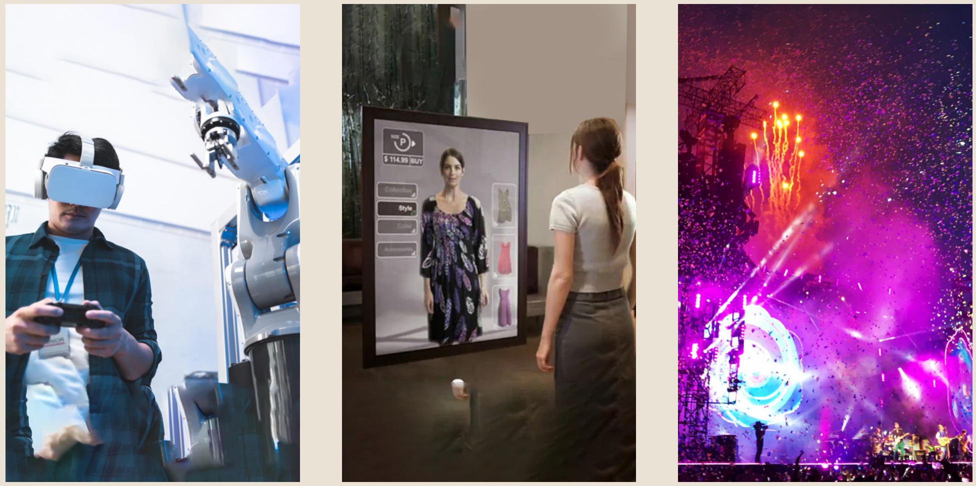
NS/EP priority services are intended to provide E2E priority from the originator of the call, over access network (radio, fixed-line, or otherwise), through the originator's home network, through any transit networks, to the destination device's network, over the terminating access network, and to the destination device. All this is accomplished using a number of different markings to the media, signaling, and user data, and through use of a number of different prioritization techniques depending on the network or technology used. The aim is to provide the highest possibility of successful service utilization (e.g., call completion for voice calls), even during times of excessive network overload (e.g., natural disasters, hurricane, earthquake, or non-natural events, sports events, road accident backups, terrorism).

Originally developed for wireline and 2G networks, support has been extended through VoIP, cable, and 4G VoLTE, and access-agnostic modern 5G networks. It is anticipated that functionality will continue to evolve to support priority treatment on future signaling and media types running on

Future Voice Network. This will require standards development alongside carrier-specific implementation to meet regulatory and customer needs.

4.2.4 STIR/SHAKEN

Signature-based Handling of Asserted information using toKENs (SHAKEN) is an industry framework for managing and deploying Secure Telephone Identity (STI) technologies for the purpose of providing E2E cryptographic authentication and verification of the caller identity in a public, IP-based service provider voice network. Caller identity information in-cludes the calling telephone number (TN), plus rich call data items such as the calling name, company logo, and caller's image. SHAKEN can also authenticate the authorized priority level for priority services such as GETS and 911 to prevent unauthorized access to communication resources during periods of congestion. At the time of writing, ATIS is completing specification work that will enable multiple countries to exchange and verify SHAKEN traffic across borders. Many future voice services may not traverse the public IP-based voice net-work, so it is likely that caller identity verification and validation framework will need to be expanded to other implementation scenarios.



Manufacturing
High-fidelity on premise analysis of machinery

Retail
Virtual try-on and smart shopping experiences

Events
Massive multi-user events with low latency streaming

Figure 1: AR/VR/XR Markets

4.3 AR/VR Use Cases

AR/VR/Extended Reality (XR) will add another dimension of depth to the calling experience. It can entertain, embed useful information into the call session, enable advertising, and facilitate better understanding between the calling parties, especially when one person is trying to assist another person remotely with a task.

AR/VR/XR applications may drive new approaches to network solutions for voice and video latency requirements. According to International Telecommunications Union (ITU) G.114 [14], voice quality starts to degrade when the mouth-to-ear latency exceeds 150 msec. For immersive visual real-time applications such as VR, multi-player gaming, and immersive visual interaction with an event or person over distance, the motion-to-photon Round-Trip Time (RTT) latency must be less than 20 msec. [15] Several factors must be considered when supporting such applications over geographically diverse end points where the delay component contributed by the network is primarily due to the speed-of-light limitation of fiber and the delay incurred as packets traverse network routers. Because the length of fiber and the number of routers increases more-or-less linearly with geographic distance, the latency itself increases linearly with distance. In addition, the regulatory and business needs may drive more anchor points in the media paths that will add delay to the communication path.

Figure 1 illustrates some examples of AR/VR/XR applications.

Descriptions of AR/VR enhancement in-call experiences include:

- While in a video call with another person(s), a discreet AR ad can overlay the corner of the screen.
- While speaking with another person(s), the subscriber can bring up an AR/VR session to explore a location such as a factory floor or provide remote training. Both parties can view the same scene together. The point of view (PoV) rendered is that of the controlling party.
- While speaking with another person(s), the subscriber can bring up a holographic session for retail.
- While speaking with another person(s), the subscriber can bring up a live event for all parties to virtually watch together.
- While engaged in a game, the subscriber can bring up an AR voice-assisted tutorial.

Future voice requirements to support these use cases include:

- Higher bandwidth and extreme low latency.
- An integrated Web Real-Time Communication (RTC) data channel in the voice client.
- Enhanced UE display may be required (e.g., for window overlays).
- The system must support secure, encrypted communications.
- Communications services must be provided in mix-and-match bundles within a single client (i.e., no siloing of voice from video from messaging from any other communications service).

4.4 Enriched Call Use Case

The UE in voice calling and messaging from communication service providers (CSPs) has been the same for decades. It is time for a major refresh. Once we have a client that can support voice as well as data services, a whole new world opens for what the end user experience can be. Figures 2.1 and 2.2 illustrate a few possibilities.

Figure 2.1: Enriched Pre-Call Experience

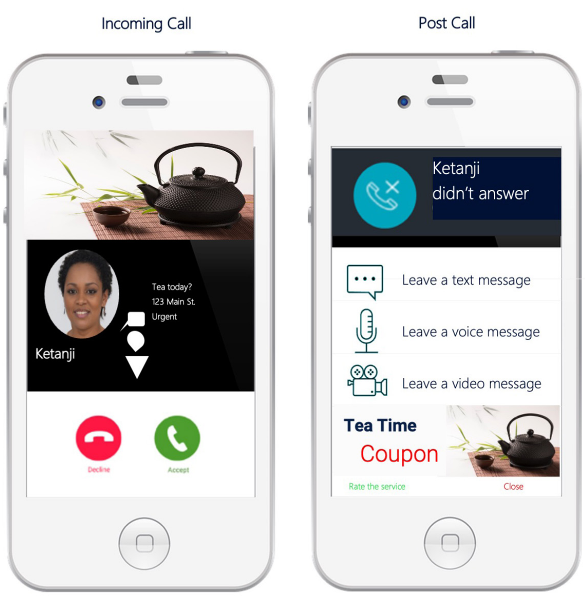


Figure 2.2 Enriched Post-Call Experience

Descriptions of enriched call experiences include:

> **Pre-call experience:** A user can “compose” information (by including a subject, location, picture, etc.) prior to placing the call such that the other person is able to see the composed pre-call information while receiving the incoming call. A CSP can add an advertisement, as in Figure 2.1.

> **In-call experience:** A user can share content during a call: chat, files (or group of files like presentations), location, background audio, video. A CSP can add an advertisement.

> **Post-call experience:** Similar to the pre-call experience, a user can “compose” additional information when a call is rejected or unanswered, for the other side to view. A CSP can add an advertisement. as in as in Figure 2.2.

> **Enriched Call Logs:** A user can see call logs with enriched information (e.g., information shared during pre-call and post-call).

> **AI enhanced content control** and/or **prescreened content library** can be used.

Future voice requirements to support these use case include:

> New interactive and adaptable clients.

> An integrated data channel (e.g., IMS, WebRTC) in the voice client.

> The system must support secure, encrypted communications with verification and integrity protection for call metadata. See discussion in Section 5.8.

4.5 Telehealth Use Case

For telehealth to be a truly viable option or to supplement in-person doctor’s visits, best-effort video calling is not sufficient. Basic video calling with high QoS is not sufficient. The doctor must be able to also use remote examination devices, such as a stethoscope or otoscope, at high quality levels for transmission of video, images, and audio. Without high QoS connections, the doctor would not be able to accurately diagnose the illness. This will also be required for the connected ambulance.

This use case is important because the world’s aging population is growing rapidly. The elderly require more frequent doctor’s visits, but they are not as easily capable of physically going to the physician’s office on a regular basis. Specialized medicine is also not available in all locations.

Figure 3 summarizes some telehealth applications to address these needs.

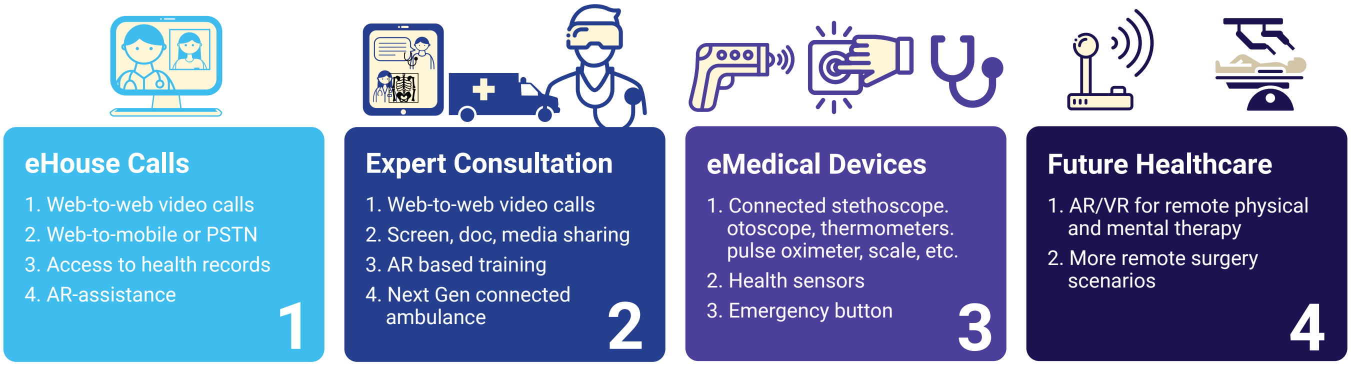


Figure 3: Telehealth Applications

Telehealth experiences include:

- **In-call experience:** Physician uses AR in call with patient or other doctors to guide the other on performing a task.
- **In-call experience:** In addition to the specific QoS for video calling with a patient, the physician will need specific QoS for use of remote examination devices such as:
  - Stethoscope – requires sufficient QoS for accurate audio.
  - Otoscope – requires sufficient QoS for accurate video.

Future voice requirements to support these use cases include:

- High bandwidth and extreme low latency.
- Sensing enabled network and devices.
- Data channel (e.g., IMS or WebRTC) to support remote medical devices and AR/VR.
- Support for real-time, non-speech audio with QoS.

5. FUTURE VOICE ARCHITECTURE

Figure 4 illustrates the future voice architecture, which will define an interoperable solution for different carriers and applications to connect users/devices together across different technologies, with voice and other media, as described in the previous section. The future voice network connects everyone in the world across those many communication layers and environments.

Key characteristics of the future voice architecture are covered in detail in the following subsections. New technologies are anticipated to bring many benefits, including virtualization, cloud native, Network as Code. Enhancements in security and trust, routing, and APIs will improve network efficiency and provide increased functionality. Dynamic service and path discovery open opportunities for new services and new types of services, many of which will have a voice component.

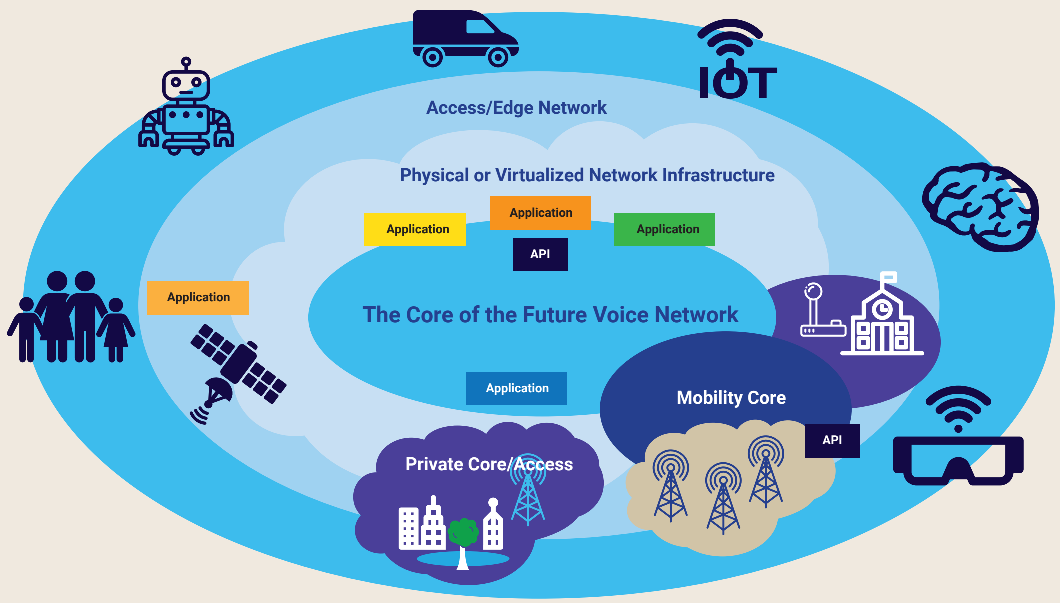


Figure 4: Future Voice Architecture

5.1 Evolution to a Highly Secure Future Voice Network Infrastructure

Network functions will continue transitioning to containerized cloud computing. These containerized network functions will benefit from the maturing of container technologies and cloud technologies, as well as the maturing of the containerized functions themselves. Network functions in support of voice will follow the larger industry path of containerized cloud native applications.

As shown in Figure 4, the proposed high-level future voice architecture includes the following components:

- **Devices/users:** The outside ring defines all devices and users, from basic voice users/devices to more advanced voice-capable AR/VR applications/devices or robots.
- **Accesses:** As an access-agnostic architecture, future voice service supports different access methods, such as wireline/fiber, satellite, mobility (e.g., LTE, 5G, 6G), Wi-Fi, or other emerging access technologies.
- **Applications:** Future voice applications provide different services/features to the users/devices. Applications are the most versatile components in architecture. They can be offered by different entities, vary in complexity, and support different devices. Applications can also reside at different layers and be embedded in either physical or virtual infrastructure.
- **Core:** The core network of the future voice architecture interconnects different carriers and accesses enabling devices/users, applications, accesses, and APIs to seamlessly work together.

A couple of key areas where significant enhancements are anticipated include multi-cluster management and security. In cloud technologies today, managing multiple clusters today is complex, involving an assortment of tools, typically needing to be customized and combined with manual procedures. Current industry trends are to use container orchestration operators to help with automation related to multi-cluster management.

Improved security is a continuous process. Any future communications network will need to address increasing threats and malicious attacks. Communications and information are of sensitive and proprietary nature. Thus, every new network deployment faces challenges such as privacy, providing customers with a secure way to communicate, operational resiliency, and regulatory compliance.

There are two primary aspects of a secure network for communications. One involves securing the customer data for privacy, sensitivity, and regulatory reasons. The other involves protecting the network itself from intrusion and attacks that can debilitate the functionality of the network's purpose. 3GPP continues to enhance network security including such areas as AI/Machine Learning (ML)-driven threat analytics for intrusion and threat management, cloud access security, and access control. Elsewhere, new approaches and enhancements to data loss prevention are under development. As new technologies are introduced to support future voice services, related security tools and capabilities will be needed for the new deployment environment and to mitigate new and existing security threats.

Trends for security improvements are focused on making security easier to manage (secret management, role-based access control (RBAC) configuration) in addition to the expected improvements to the overall security posture of containerized applications.

Containerized network functions are also expected to continue to evolve into fully cloud native applications, thus reaping the benefits of cloud scale and performance. A fully cloud native architecture will allow network functions to be deployed seamlessly either in on-premises clouds or utilizing public clouds, whichever makes the most business sense.

Even as Network Functions continue their transition to containerized cloud computing, exceptions will exist where dedicated non-cloud physical infrastructure can be beneficial. It may be driven either by security, business functions, or the application functions where dedicated non-cloud hardware is more applicable (e.g., transcoding). The integration of the cloud and physical environment is required.

5.2 Core Architecture and Protocols

IMS is a VoIP-based architecture using SIP to provide voice and multimedia services to everyone. The Core Session Control Function (xCSCF), often referred to as the "IMS core," provides an infrastructure where xCSCF network functions "loop" signaling through the application servers for voice/video, text, RCS, 3rd party apps, etc., regardless of access network types. This is controlled by a consolidated subscriber profile repository, which identifies the application servers needed for that subscriber for a requested service session.

The IMS Core design offers a flexible solution for service and feature development. It also avoids a proliferation of "point solutions" (a.k.a. "service islands" or "walled garden"), which make service interoperability difficult. However, the IMS architecture is considered complex. The IMS Core is compute intensive, with multiple distinct CSCF functions, each performing a different part of call processing tasks. For example, the S-CSCF network function routes signaling one by one through application servers to ensure service interoperability. Additionally, the SIP protocol, as a session-aware and connection-oriented protocol, is often considered processing intense and heavy. Accordingly, many industry initiatives have started to investigate how to simplify IMS or offer a lightweight IMS but still maintain a truly unified core network architecture. Other initiatives are investigating how to use other protocols (e.g., HTTP2) that may be lighter than SIP to provide similar or better call control function.

Proposed optimizations for the session control function include limiting the functionality to controlling connections, pushing media management to the endpoints, and consolidating the multiple CSCF types into a single one.

In either case, it is expected that session control and media control will remain separated to enable meeting the performance requirements of the media path (e.g., voice, video, AR/VR). Another optimization for IMS could be enhancements to use the 5G unified and federated databases. Doing so could open opportunities for simplified authentication, policy control, and application management.

Some of these initiatives are also looking at IMS deployment considerations. The advent of virtual infrastructure enables easy scale up/down of resources as business needs change. This flexibility makes feasible implementations suited to customers of varying sizes and enables add-on solutions, such as MEC data applications, that are particularly relevant to enterprise environments.

5.3 Network as Code Enabled by APIs

The ability to easily develop new applications integrating in disaggregate communications services, along with the ability to quickly assign different network QoS levels for applications, is important for future voice services. Network as code is an approach to simplifying the network capabilities such that they become part of distributed service chains while maintaining secure access to exposed network functions. These service chains optimize mission-critical functions locally while providing global access at scale.

For example, a transport application may use a service chain encompassing location services, a hosting cloud, and traffic database. The service chains can be viewed as software development kits (SDKs) that enable developers to use APIs exposed by the network and operation services to create higher level applications. Using these SDKs means application developers do not require specialized network or operations knowledge to create applications requiring assured connectivity. The APIs provide access to resource-specific service level agreements (SLAs) and support the application developers by providing control functions at the level the developer is familiar with.

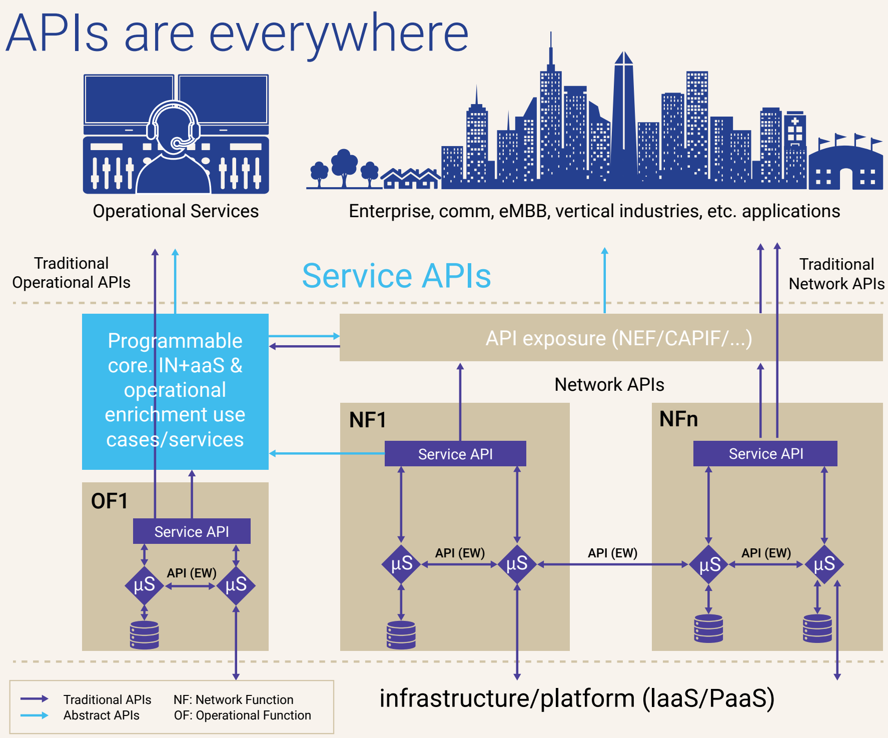


Figure 5: APIs for Network as Code

Although network-based APIs have been available for some time, they offered limited functionality that required application developers to have a significant level of network knowledge to create useful applications. SDKs available with Network as Code provide confidence to the network provider that the exposed network functionality is secure from unauthorized access. At the same time, the Network as Code experience allows a developer to have secure access to:

> Communications services APIs.
> Network APIs, including for QoS and sensing.
> Operations service and SLA APIs.
> Data framework to enable data producers to share data with data consumers (e.g., AI applications).

Network as code is designed with ease of application development and programmability in mind, while also ensuring network security. This difference will enable greater access to and use of APIs by new and diverse developers from the network operator, enterprise, and OTT communities. This opens possibilities for new applications and services such as those described elsewhere in this document.

5.4 Dynamic Over-the-Top (OTT) Application Access via IMS Data Channel

Future voice service introduces an opportunity to overcome the OTT island limitation, increasing opportunities for interoperability while also providing the benefits of OTT applications in a carrier-controlled environment. The IMS data channel being introduced in 5G Advanced standards provides a general mechanism to support UE access to multiple applications, without those applications having to be pre-subscribed or pre-downloaded to the UE. A UE can request access to a specific application via the IMS Data Channel (DC) as needed and extend the use of the same application to other UEs in the same IMS session.

This dynamic app access avoids the need for having many apps on a UE to communicate with various other friends, family, and businesses, each of which may favor a different OTT communications app. The user can dynamically request an IMS DC session with, for example, a specific messaging app to communicate with a family member who prefers that app and later use a proprietary IMS DC physician's communications app over the IMS DC with their doctor. Imagine the user being able to use either a hyperscaler OTT app or business-specific apps over the IMS DC with any person or business (e.g., auto repair shop, bank, pharmacy, restaurant, or hotel) without requiring the user to pre-subscribe to or pre-install the app.

Figure 6 illustrates the call flow. After the IMS bootstrap DC has been established, either party can update the session to use an app from the IMS DC repository. In the example below, party A initiates the request to use the app. After both parties accept, the IMS DC path is established for both parties to access the same app from the repository.

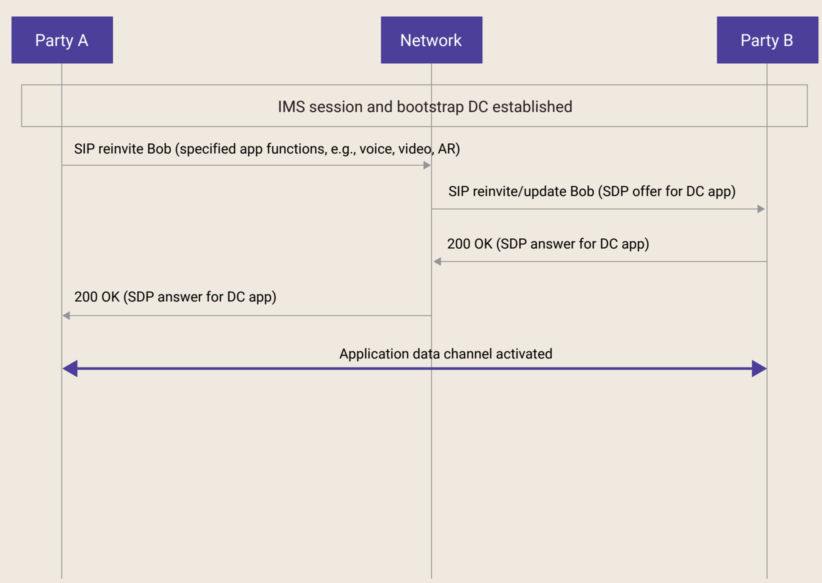


Figure 6: IMS DC Bootstrap Call Flow

The application may be an OTT app that has been stored in the IMS DC repository, perhaps with some modification for this type of use. In this case, the app may be selected in this manner after a negotiation has determined no applicable common app between the end parties. In another example, the app could be an enterprise-specific app provided by a dentist, hotel, or other business to provide customized services for client interaction. In any case, an app designed for this type of IMS DC access could be accessed without either party having to have the app pre-installed into their UE.

5.5 Service/Application/Capability Discovery and Interoperability

An application-discovery process could be introduced to the IMS DC to negotiate the application capabilities/preferences between two or more parties. After a handshake to share information about endpoint capabilities and preferences, one or more applications may be selected, and communication channels created to allow the end users to have feature-rich communication using their preferred application(s).

The users and end devices have full control of the application preferences. Preferences can be dynamically modified based on each user's needs and device capabilities. The application capability preference can also be further defined by the users to include their own application proprietary parameters.

It is envisioned that this flexibility in discovery and application selection allows the end users to negotiate a feature-rich application based on pairwise preference and capabilities. Embedding this flexibility in the IMS DC allows traditional telco providers to offer interoperability to other application providers. Following a successful negotiation, the telco provider can manage QoS for the connections, update the applications, or provide other support. If the negotiation fails to find a suitable application, the telco operator can provide some combination of basic voice and video support.

5.6 Addressing and Interoperability

Until now, basic voice and messaging services were truly interoperable services. To universally identify the voice service endpoints, the hierarchical and pseudonymous decimal-number-based scheme described in E.164 has been used for interoperable addressing. E.164 generally assumes that numbers are allocated by national regulatory bodies or similar. However, there are some exceptions to this (e.g., global satellite services, non-geographical allocation, 800 service numbers, and some mobile number overlays). E.164 was developed in the early era of circuit-switched switches and has provided the industry with backward compatibility as it moved through a gradual transition to packet voice services.

Currently, the E.164 numbering plan is still widely used on most networks, including all IMS-based VoLTE networks, most Machine to Machine (M2M) communication, and some IoT devices. However, with the exponential growth of IoT devices and M2M communication, E.164 numbering plan does not have enough address space to support all applications/devices. Additionally, deployment of direct VoLTE interworking has been limited and based on E.164 numbering to date.

Over the past 10 years, there has been much innovation and introduction of new voice-based communication services. However, these predominately do not use E.164-based identifiers for routing their users' communications, are mostly incompatible, and either have minimal interworking or no interworking at all.

For future voice service to become the dominant communication method of the next generation, it will need to leverage non-E.164-based routing methods. For example, SIP addresses may be used with names and domains instead of numbers, and a prefix to identify the destination (e.g., firstname.lastname@carrier.com, nickname@host.com, lastname-initial@employer.com). Although telephone universal resource identifiers (TEL URIs) are used in IMS today, and while they are a form of SIP address, they use a fixed formatting focused on the E.164 number as part of the address.

Given its dominant adoption within the industry, IMS will probably be the baseline for future voice services. Therefore, IMS will need to evolve to accommodate non-E.164 means of addressing subscribers and routing services. To achieve this, several enhancements are anticipated. Increasing use of DNS-based routing allows flexibility to locate subscribers anytime, anywhere, and on any device. This can include a non-TEL URI-based addressing scheme. Number, or identity, portability will continue to be required, regardless of the addressing and routing scheme in use. From an interworking perspective, support will continue to be required across different carriers and different identity schemes. E2E service control and routing will also be requirements.

5.7 Inherent Voice Support

When cellular networks were first developed, voice services were integrated with the core network. The most recent generations have separated the voice networks from the underlying packet network, which accordingly has many implementation complexities, architectural variations, and interworking issues, resulting in deployment delays and costs. This has necessitated solutions to hasten voice support, which have predominantly incorporated some form of fallback to earlier underlying voice networks. While hastening support for voice, fallback has many issues that delay new service adoption. These include:

- New generation services are not available while a voice call is in progress.
- Mixed support for voice by devices and networks confuses users who don't know if the person they are "calling" supports the new service they are attempting. This results in slow take-up and a Catch-22 situation.
- Legacy equipment and spectrum allocations must be kept in service just to support fallback.

To avoid potential service gaps resulting from a lack of voice service networks and devices supporting future voice service, the future network should inherently support voice from the start, even if a phased introduction of services is planned. Network operators will need to support existing voice coincident with the launch of future voice services and the future voice networks. This will include all regulatory requirements and services related to voice. Thus, all voice devices capable of using the future voice network will need to support existing voice services from the very start, even when few, if any, future voice networks have advanced capabilities. Indication of future voice service support should occur only if both the network and device are future voice service capable.

Any required support services or documentation to enable the support services will need to be in place to support transition well prior to the availability of the first commercial networks supporting future voice service. This includes sufficient information to allow development of supporting back-office systems (including billing and roaming). Thus, the industry must define and implement capabilities to allow internetwork roaming. This will need to support a scenario where devices capable of future voice service can roam using the future voice service, even if their home networks does not support this service. Although fallback functionality may be necessary, it should be used only as a last resort to accommodate future voice network failure or other disaster scenarios.

5.8 Trust and Security

The future voice network presents a number of security challenges across all the dimensions of security associated with providing and using voice services and voice-based applications. This is in addition to security of the network infrastructure as described elsewhere in this document. These include the dimensions of identity, authentication, access control, confidentiality and privacy, integrity, non-repudiation, and availability among the various users and service providers involved in voice sessions.

Related to these dimensions is the establishment of trust on bilateral and multi-lateral bases across the ecosystem. This is the process of determining which entities in the ecosystem (e.g., human and machine users, service providers, and applications) are expected to follow policy norms to make voice service connections, exchange information, and transact business over those connections. Trust encompasses the presentation of identity and trust-characteristic information between users, service providers, and 3rd-party information sources. The following are a few key characteristics of the anticipated future voice network and the challenges they raise to some security services and trust characteristics.

Disaggregation of the Device, Data Channel, Voice Application, and VoIP Service

Newer-generation voice networks may have different permutations of device, app, credential provider, data channel provider, and voice service provider. This means the credentials for authenticating into a voice service may be separate from what the device uses to authenticate with the data channel provider. The data network path to the voice service provider may also extend over the public network. As a result, the data channel provider is only partly responsible for the availability of that path, and the public network is inherently untrusted and may require additional security services (encryption, denial-of-service protections, etc.) between the application and voice service provider. The voice service provider may also need to engage in its own identity proofing and device/application registration process with the user separate from account establishment and registration with the data channel provider.

Migration of Network-to-Network Interfaces to the Cloud

With IP infrastructure migrating to the cloud, the inter-carrier Network-To-Network Interconnect (NNI) interface between carriers will change from dedicated IP network cross-connects/static network configurations to cloud-based dynamic virtual interfaces. This change introduces new security requirements such as determining the identity of a cloud-based peer, authenticating that peer's network functions, establishing security associations for confidentiality and integrity using cloud-resident credentials, and enforcing data flows and service levels. It may also add new policy requirements such as virtual private network (VPN) or application-level authentication, methods of enforcing signaling and media access control, and packet flow policing.

Increased Real-Time Communication Between Human Users and Non-Human Applications

An increasing amount of voice communication occurs between human users and non-human business applications that are gaining more sophisticated conversational voice capabilities. Although significant progress has been made in support of establishing trust relationships between human parties on a voice call, there is still room for improvement. The introduction of non-human voice applications to voice calls adds another dimension to the establishment and assurance of a trust relationship between the parties. New or enhanced methods of verifying the identity of a party — as well as for sharing trust information, intent, and purpose of the communication — need to be investigated to address both current and future gaps in the trust relationship.

5.9 Access Agnostic

Future voice service offerings and service requests are abstracted so network implementation can be configured in any number of ways, even across multiple providers. It should provide a converged experience across multiple access technologies and service scenarios, including wireless and wireline networks to ensure a consistent consumer experience.

The future voice architecture should be inclusive of emerging services and innovation in applications. If the access network can support the required key performance indicators (KPIs) and has an agreed SLA, the access and edge network need to support service porting from one access to another so that user devices/experiences can continue in all operating scenarios.

With the newly introduced technologies, the access and edge components of the network should also support adequate federation and sharing. Critical services, Open APIs, and sharing of computing networks will add to the need for new security and privacy solutions at the access and edge layers.

6. AN EXAMPLE OF FUTURE VOICE ARCHITECTURE AND SERVICES

The following is a comprehensive example of services that could be supported by future voice architecture.

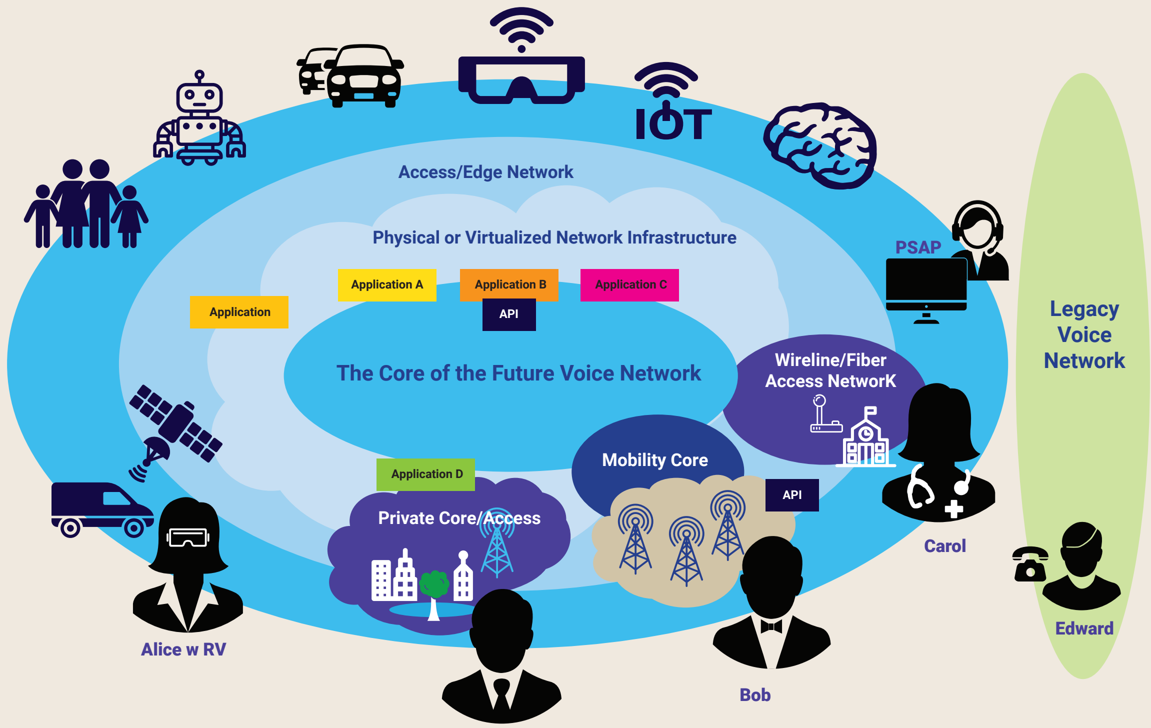


Figure 7: An Example of Future Voice Network

The main character in the above example is Alice. She is traveling with her high-tech RV from place to place and keeps connected with her work and with her family and friends. Her job as a travel guide is to travel to different national parks and provide real-time AR/VR virtual tours to their customers.

Alice has the access to the following devices and accesses:

- Her high-tech RV:
  - Provides connectivity for Alice if she is on or around her RV.
  - Can be intelligently connected to any type of access network.
  - Equipped with IoT sensors for telehealth.
- Her AR/VR set.
- Her mobile device can support multiple accesses (e.g., satellite access and regular mobile access).
- Her IoT sensors for telehealth application.

Alice can do every application if her access network allows, including AR/VR, immersive telepresence, telehealth, voice, video, messaging, etc. Alice may use a super app to manage all her application needs.

Alice's main communication needs are:

1. Contact people on her contact list that include:
   a. Alice's family member Bob, who is served by application/service provider B that offers general mobility services.
   b. Alice's doctor Carol, who is served by application/service provider C that offers telehealth service with IoT sensing capability.
   c. Alice's travel agency manager David, who is served by application/private core D offers advanced real-time AR/VR services to its customers.
   d. Alice's other brother Edward, who has lost his smartphone.
2. Contact other people for other purposes:
   a. Being contacted to perform a research study offered by a human behavior research company.
   b. Book an RV park.

The following are some assumptions, basic features, and requirements offered by the core of the future voice network:

- All users can be served by different service providers and applications.
- There should be trust and authentication between users to the different networks and among different layers of networks and applications.
- All different providers would be able to communicate with a user via the user's global routable identity.
- Among service providers, connections to interconnect arrangements are established between each other or via 3rd-party carriers.
- All applications can use open APIs to request and negotiate QoS with their access service provider (e.g., mobility core service provider, fiber access provider) based on the access SLA arrangement.

The following use cases will utilize the application discovery process described in Section 5, with IMS or other protocol as the control signaling, where it can dynamically invoke or update communications based on required QoS for individual applications/sessions.

1. AR/VR calls with immersive telepresence sessions from Alice to her manager David to provide a virtual tour experience to her customers:

   Alice uses her application D to contact David to have a pre-arranged AR/VR session with their customers at David's corporate location. Using a future voice service, they establish the connection and use the IMS Data Channel (controlled by IMS or other control signaling) to activate an immersive telepresence session provided by David's travel agency. After the initial presentation using telepresence, it is time for all customers to have a virtual experience in a national park using their AR/VR headsets. Alice uses the future voice service control channel (IMS) again to update the data channel connection to support AR/VR. Alice then will lead an AR/VR-enabled virtual tour of the national park.

2. AR/VR call from Alice to his family member Bob when access and QoS allows:

   Bob calls Alice while she is traveling in her RV. When Bob calls Alice, he is using the future voice service application discovery capability, which determines that both Alice and Bob can use Application B to communicate. When the call starts, both Alice and Bob have abundant bandwidth to establish AR/VR connections.

3. Voice call from Alice to Bob when Alice only has limited bandwidth:

   Continuing from the above use case, while the communication session continues, Alice is driving into a remote location in the mountains with low-bandwidth cell coverage. When the future voice service detects this, the connection's control channel initiates an update session to modify the AR/VR session to a voice/video session and later drops to a voice call when the coverage continues to degrade. As Alice parks her RV and starts hiking, her communication via RV is switched seamlessly to her mobile device, which also has satellite capability. After she hikes for a while, Alice is no longer able to receive any cell coverage, Alice's device automatically enables the satellite connection. Alice can continue her voice communication using satellite without interruption.

4. Voice call with telehealth from Alice to Carol when access and QoS allows:

   Alice is on the road most of the time of the year, so she has established her routine checkups and sick visits using her telehealth service. Her RV is equipped with sensors for telehealth. Alice is on her scheduled virtual well visit with her doctor Carol. Alice makes a basic video call to Dr. Carol in her RV with a request to enable a telehealth application. The application discovery process routes the call to Application C, which supports telehealth applications. Dr. Carol responds. After the initial conversation, Dr. Carol and her nurse enable the telehealth sensor application. Under Dr. Carol's voice and video instruction, Alice connects to the sensors equipped in her RV. Dr. Carol initiates the telehealth bio exam and updates the data channel connection to support the QoS required for sensors, such as a stethoscope and otoscope. All of Alice's bio-data and exam data will be continuously transmitted to Dr. Carol.

5. Booking an RV park using AR/VR-enabled telepresence application:

   Alice is planning to book an RV park for her next trip. She first establishes a basic connection with the RV park using the RV park's global identifier. After the connection is set up, with Alice's permission, the RV park's application is invoked using the IMS data channel. Alice then uses the RV park's application to enable an AR/VR virtual tour of the RV park facility. During the connection set up, a modified QoS is established at Alice's end to support the QoS session of the AR/VR connection for a virtual tour.

6. QoS and service discovery use case:

   Alice and Bob are on a regular voice/video call and decide to accept a request from a health/behavior research company that performs health behavior analysis for gamers. They start a call on regular voice/video, then add a game using the latest AR/VR capability running on a special application. The control channel initiates an update of the QoS needed for the game and AR/VR applications. At the same time, they establish connections to allow the research company to collect their bio-status remotely using multisensory communication. All QoS requirements can be controlled by the initial voice/video control signaling (e.g., NEF, DC signaling, or 3rd-party call control by the research company E). This QoS arrangement can also be updated mid-call per user's preference or based on the update of the application, the environment, or other factors (e.g., different applications) using the voice establishment signaling as control signaling for all applications.

7. Emergency call (e.g., 911 case):

   Alice gets lost on her backpacking hike. She makes a 911 call using her cell phone. The future voice service routes the call to an upgraded PSAP. Based on Alice's location, the PSAP uses the enhanced call capability to send directions to Alice. Alice follows the directions. However, at some point, the cellular signal gets weakened and the 911 call is switched to satellite. While maintaining the connection, Alice loses her enhanced call capability. The PSAP continues to guide Alice to a safe place via voice only.

8. Interworking with legacy voice users/networks:

   Edward has lost his future voice device. He contacts his service provider to request a replacement. Because he is using an old telephone, he asks his service provider to temporarily associate his future voice service account with the phone number of this old telephone even though it does not support future voice functionality. Realizing he will not be able to participate in a weekly telepresence call with his sister Alice, he calls her to let her know. The old telephone does not support downloaded applications (or enhanced voice applications). So Edward uses the legacy phone interface to place a voice call, which is routed to Alice's future voice device. After the call is over Alice remembers that she needs to remind Edward that it is Bob's birthday this week. Alice enters Edward's normal contact information into her future voice device. This call is routed to the temporary phone number (E.164) that Edward is using, and a voice-only call is established to Edward's old telephone.

The above example demonstrates the use of the following future voice capabilities:

- Establish communication connectivity with anyone.
- Interconnect between different layers of application/communications offered by multiple carriers and networks.
- Support application discovery and real-time invocation of services.
- Interrogate QoS requirements from the application to the access network.
- Real-time control, update, and adapt connections' QoS needs based on the application that is running.
- Seamless handover among different accesses.

7. CONCLUSIONS AND RECOMMENDATIONS

There will be different deployment models for voice services. The blurring of voice services and other services made possible by IP-based cloud applications will become more apparent. Future voice applications running on IP clouds and over IP networks will have different levels of integration with the underlying data connectivity and different levels of interworking with the public voice network.

The future voice architecture needs to accommodate the full range of future voice applications. In this report we have identified the following features as important pillars:

- Support for basic and regulatory voice services should be a native part of future mobile standards from Day 1 to mitigate the problems caused by fallback to previous generations.
- IMS has been an important technology to modernize carrier voice support but is now mature. The industry should assess whether IMS can be enhanced to meet the needs of the future voice architecture.
- Voice services should offer a high degree of privacy, trust, and security. Establishing a trust relationship between the parties by including verification of identity and sharing trust information, the intent, and the purpose of the communication will be critical.
- The future voice architecture should support rich and interoperable voice services both within and outside the public voice network. The future voice architecture should enhance innovation in the public voice network and expand its service capabilities to offer increased value to service providers.
- The technical implementation of the future voice architecture should be characterized using cloud native concepts enriched by APIs to expose network functions to support enhancement, evolution, and customization of services.

This report recommends that 6G mobile standards include a future voice architecture that addresses the requirements for efficient and interoperable voice services identified in this report.

ACRONYMS

2G	2nd Generation Wireless technology
3GPP	3rd Generation Partnership Project
4G	4th Generation Wireless technology (3GPP)
5G	5th Generation Wireless technology (3GPP)
AI	Artificial Intelligence
API	Application Programming Interface
AR	Augmented Reality
CALEA	Communications Assistance for Law Enforcement Act
CAPIF	Common API Framework for 3GPP northbound APIs
CN	Core Network
CSCF	Call Session Control Function
CSP	Communications Service Provider
CT	Core Terminals
DC	Data Channel
DNS	Domain Name System
E2EE	End-to-End Encryption
ESInet	Emergency Services IP network
FF	Factories-of-the-Future
GETS	Government Emergency Telecommunication Service
HSS	Home Subscriber Server
HTTP	Hypertext Transfer Protocol
IETF	Internet Engineering Task Force
IMS	IP Multimedia System
IoT	Internet of Things
IP	Internet Protocol
ITT4RT	Immersive Teleconferencing and Telepresence for Remote Terminals
ITU	International Telecommunications Union
IVAS	Immersive Voice and Audio Service

KPI	Key Performance Indicator
M2M	Machine to Machine
MC	Mission Critical
MEC	Mobile Edge Computing
MIMI	More Instant Messaging Interoperability
MIoT	Massive Internet of Things
ML	Machine Learning
MMES	Multimedia Emergency Services
MPS	Multimedia Priority Service
Msgin5G	Message Service in 5G System
NENA	National Emergency Number Association
NGCS	NG911 Core Services
NIDD	Non-IP Data Delivery
NS/EP	National Security or Emergency Preparedness
OTT	Over the Top
PCF	Policy Control Function
PLMN	Public Land Mobile Network
POV	Point of View
PSAP	Public Safety Answering Point
PSTN	Public Switch Telephone Network
QoS	Quality of Service
RAN	Radio Access Network
RBAC	Role Based Access Control
RCS	Rich Communications Services
REST	Representational State Transfer
RTC	Real-Time Communications
RTT	Round-Trip Time
RV	Recreational Vehicle

ACRONYMS

SA	System Architecture
SBA	Service Based Architecture
SBC	Session Border Controller
SDK	Software Development Kit
SEAL	Service Enabler Architecture Layer
SHAKEN	Signature-based Handling of Asserted information using toKENs
SIP	Session Initiation Protocol
SLA	Service-Level Agreement
SMS	Short Message Service
STI	Secure Telephone Identity
TEL URIs	Telephone Universal Resource Identifiers
UAS	Uncrewed Aerial Systems
UE	User Equipment
URI	Uniform Resource Identifier
V2X	Vehicle to Everything
VoIP	Voice over IP
VoLTE	Voice over Long Term Evolution
VPLMN	Visiting PLMN
VPN	Virtual Private Network
VR	Virtual Reality
WPS	Wireless Priority Service
XR	Extended reality

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